Profile: PHILIP GLASS

MODERN RECORDING & MUSIC

47947
$1.95

MARCH 1983
VOL. 9 NO. 3

Spyro Gyra

DING TECHNIQUES,

RTS:
50 Reel to Reel
ar Pro 700 Amplifier
3000 Delay Line

NEW PRODUCTS
RECORD REVIEWS

WE JUST BROUGHT SMPTE CONTROLLED AUDIO EDITING UP TO SPEED.

End slow audio interlock with our Series 50 compact recorder/reproducers. They're designed for demanding production applications like SMPTE control. Built smarter and stronger, with better electronics to make interlock and editing faster.

All three motors—the strongest we've ever put into a compact unit—are included in the microprocessor-controlled servo system for faster and smoother tape handling. Input from five independent photo interrupters and multiple tachs provide a more accurate detection of tension, tape speed and direction, improving external computer control. A single connector delivers all status and remote signals for editor/synchronizer interface and eliminates complicated wiring.

For non-sync recording, two built-in auto locators find critical cues with a single command. Tape location is presented in minutes and seconds, positive or negative.

For easy maintenance and service, all electrical and mechanical adjustments are made from the front and each audio channel is built on its own glass epoxy PCB.

The Series 50 recorder/reproducers are tougher and work harder than any compact machines we've ever produced. Tough enough to bring your audio editing up to speed.

For more information see your TASCAM dealer or write us at 7733 Telegraph Road, Montebello, CA 90640, (213) 726-0303.

TASCAM

TEAC Production Products

DESIGNED BY POPULAR COMPLAINT.

CIRCLE 10 ON READER SERVICE CARD

MARCH 1983
VOL. 9 NO. 3

MODERN RECORDING & MUSIC

THE FEATURES

RECORDING TECHNIQUES—PART XI 12
By Bruce Bartlett
This month *MR&M* looks at the monitor system, which provides the information necessary to judge what's just been recorded. The roles of speakers, crossovers, and room acoustics are examined.

RECORDING WITH SPYRO GYRA 22
By Amy Gordon
MR&M spoke with Jay Beckenstein and Rich Calandra about Spyro Gyra's latest work, and with Michael Barry, their engineer. Here's the chance to read what kind of gyrations went into their latest album, *Incognito*.

PROFILE: PHILIP GLASS 33
By Allan Kozinn
Philip Glass, the avant-garde composer whose first opera was *Einstein on the Beach*, talks to *MR&M* about the Philip Glass Ensemble, and some of his creative works, including LP's, opera, and a filmscore. Kurt Munkacsi, the ensemble's engineer, talks a bit about the recording process.

COMING NEXT ISSUE!
A Session With Graham Nash
Profile: Lester Bowie
The Augmented Effectron

Cover Photo: Rod Taylor
Jay Beckenstein Photo: Rod Taylor
Spyro Gyra B&W at board: Dennis Gore
Philip Glass & Kurt Munkacsi Photo: Paula Court

THE STAPLES

LETTERS TO THE EDITOR 2

TALKBACK 6
The technical Q & A scene.

THE PRODUCT SCENE 16
The notable and the new, with items from Orban, Teac, Nakamichi, and others.

AMBIENT SOUND 46
By Len Feldman
Once again, digital is deftly defended. In the words of another great writer, it seems that not only will digital survive, but it will prevail.

NOTES 41
By Craig Anderton
The Korg SDD-3000 Delay Line.

LAB REPORTS 48
By Len Feldman
Otari MX5050 Reel to Reel
Phase Linear Pro 700 Amplifier

HANDS-ON 56
By Michael Tapes
Otari's 5050 Reel to Reel

GROOVE VIEWS 59
Reviews of albums by Joni Mitchell, Grace Jones, George Russell and John Lewis.

ADVERTISER'S INDEX 63

Modern Recording & Music (ISSN 0273-8511) is published monthly by MR&M Publishing Corp., 1120 Old Country Rd., Plainview, NY 11803. Second class postage paid at Plainview, New York, and at additional mailing offices. Subscription rates: $15.00 for 12 issues; $26.00 for 24 issues. Add $3.00 per year for subscriptions outside of U.S. Subscriptions must be paid in advance in U.S. funds. Postmaster: Send Form 3579 to MR&M Publishing Corp., 1120 Old Country Rd., Plainview, NY 11803.

MODERN RECORDING & MUSIC

JOHN WORAM
Editor

GENE SCARAMELLINO
RICKI ZIDE
MARK WALDSTEIN
Associate Editors

LEONARD FELDMAN
BRIAN ROTH
Technical Editors

CRAIG ANDERTON
ROBERT ANGUS
VICKI GREENLEAF
NAT HENTOFF
STAN HYMAN
FRED RIDDER
JAMES RUPERT
Contributing Editors

ELLEN ZOE GOLDEN
ROBERT HENSCHEN
JOE KLEE
ALLAN KOZINN
STEVE ROW
JEFF TAMARKIN
Music Reviewers

KAREN COHN
Advertising/Production

KATHI LIPPE
Layout and Design

SPARTAN PHOTOTYPE CO.
Typographers

BERNADETTE WEBER
Circulation

LARRY ZIDE
Publisher

Editorial and Executive Offices
MR&M Publishing Corp
1120 Old Country Road
Plainview, NY 11803
516 433-6530

Editorial contributions should be addressed to The Editor, Modern Recording & Music, 1120 Old Country Road, Plainview, N.Y. 11803. Unsolicited manuscripts will be treated with care and must be accompanied by return postage.

Letters to the Editor

A Letter *From* MR&M

Sharp-eyed readers may have spotted a few changes in this little magazine recently. In particular, the masthead next to the Letters to the Editor column appears to have been set in some sort of movable type. It seems that every month there's a different cast of characters there.

If you read the fine print at the bottom, you may have discovered that *MR & M* was recently purchased by the publishers of **db—The Sound Engineering Magazine**. That accounts for the editorial musical chairs that have been going on, as the staffs of the two magazines were being coordinated.

Notice that we didn't say "merged." Unlike National Airlines, which disappeared into Pan Am, *Modern Recording & Music* will retain its own identity, and continue to serve the audience it has been serving all these years.

Who's Out There?

Judging by the letters we get, we think we have a pretty good idea of who you are and what you want to read about. Here's how it looks to us. (Let us know if the description fits. We also do custom alterations at no charge.)

You're very interested in learning everything you can about...modern recording—and music. (That reminds us of the name of a magazine we once saw somewhere.) Some of you are more comfortable with a soldering iron than with a guitar. For others, it's vice versa. In other words, you may approach the music from different sides of the control room window. But you all want to get it down on tape in the best manner possible.

Hi-fi is OK, but that's not really what *MR & M* is all about. Of course, you'll need some sort of fi (hi, or otherwise) later on, but here we're more interested in the recording system. We think you are too.

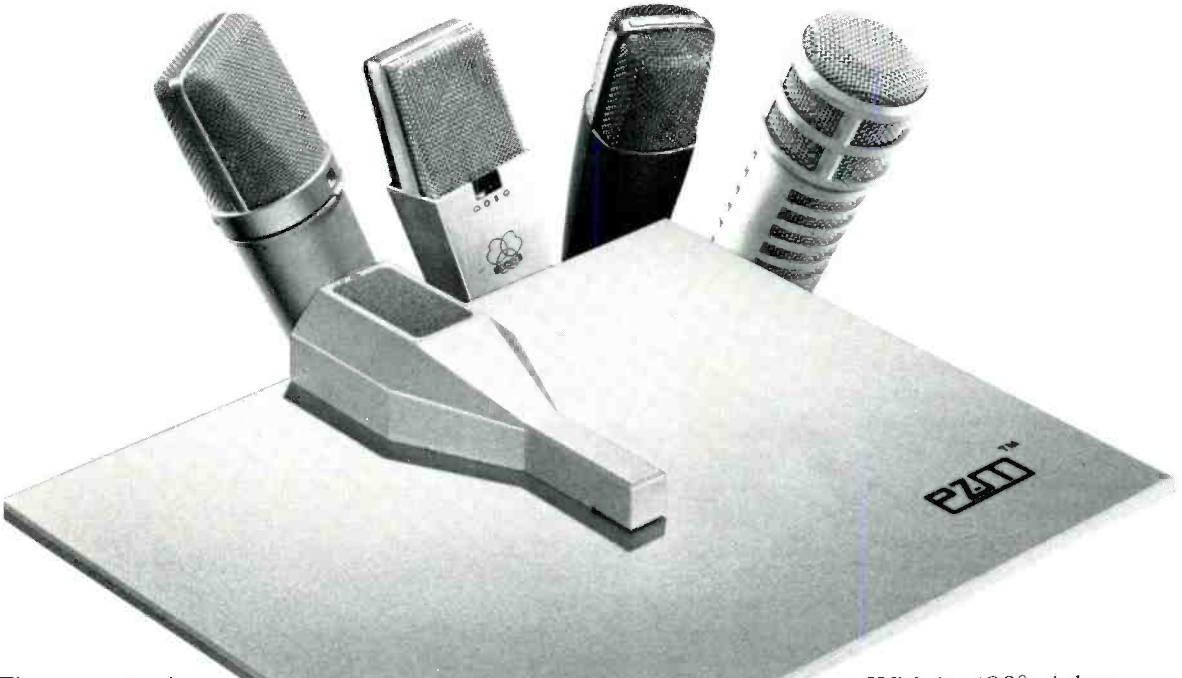
Is recording your hobby or your profession? Chances are, most of you started out as amateurs. (Doesn't everyone?) Some have already gone—and others are planning to go—into full-time recording. Still others look at the whole recording scene as a rewarding hobby, or as a part-time occupation on weekends.

You want the best equipment you can afford (nothing new about that), and some help with getting it to behave. If you're a musician first, you probably don't need us to help you with your performance skills, which is a good thing considering our skills (non-existent) in this area. (The last time we picked up an instrument, the only request we got was to put it down again.)

However, we're a little more comfortable with getting those instruments on tape, somewhere between the noise level and saturation. So if you're a musician, we can help you cope with the complexities of the control room. And if you're more at home on our side of the glass, we can help there too. Stick around and you'll see how.

JMW

STAR PERFORMERS



The new star is the Crown PZM® microphone, rapidly taking a leading position beside past favorites with quality-conscious recording studios and sound engineers. And for good reasons.

The PZM offers extraordinary reach and clarity, providing new latitude and simplification in miking setups. For drum pickup, for example, an overhead PZM, plus a ribbon or condenser mike, is often all that is needed.

Crown PZM enhances recording quality, as well. A skeptical symphony conductor, after recording with only two overhead PZM mikes, enthusiastically proclaimed that it was the first time he had heard on tape what he heard on the podium.

With its 180° pickup, accurate 30-foot range and no off-axis problems, Crown PZM is becoming the microphone of choice for recording theatrical productions. And its low-profile, hidden look makes the PZM ideal for lecterns and TV.

Try the Crown PZM. It could help you earn a star performer rating for your next project.

FREE PZM NEWSLETTER

Add to your PZM education with PZMEMO, the Crown newsletter of PZM application ideas. For your free subscription, simply mail this coupon to Crown. Do it now because PZMEMO will soon announce a new, totally redundant clip-on PZM you will want to know about.

Name ______________________

Address ______________________

City ____________ State ________ Zip ________

crown®

...WHEN YOU'RE READY FOR REAL!

1718 W. Mishawaka Rd., Elkhart, IN 46517, (219) 294-5571

CIRCLE 13 ON READER SERVICE CARD

Can You Hum a Little Harmony?

I am interested in becoming a recording engineer, but have one problem (I think). I have no musical ability. I am currently working on an A.A.S. in Electrical Technology, and have been involved in audio as a hobby for several years. I am sure I could handle the technical end of the profession.

How necessary is it for a recording engineer to have musical abilities? I enjoy music very much, and consider myself fairly creative, but have never been able to play a musical instrument (except by rote memorization). Is it worth a try? Your comments would be greatly appreciated.

—David Weinbach
Rochester, NY

If you're planning a career as a recording engineer, don't be discouraged by a lack of ability as a musical performer. Recording technology shows no sign of slowing down, and you'll have all you can do to keep up with the technical side of the profession. What counts is how well you perform on the control room side of the glass, and the professional recording system is a formidable instrument to master. Learn all you can about playing (but not playing with) the console, and let the others make the music.

Incidentally, many recording studio owners are getting to the point where they may be ready to consider kidnapping talented engineers who can demonstrate an ability to keep the equipment operational. If you have any interest and skill in this area, it's worth a lot more than the ability to strum a I-IV-V chord. Good luck!

The Five Sounds of Roland

Regarding a Talkback letter you printed in the December, 1982 issue of *MR&M*, how about using a Super TR 606 Drumatix?

The Roland TR 606 can be easily modified to have separate outputs for each of these five sound: 1) Bass drum, 2) Low and high toms, 3) snare, 4) cymbal, 5) open and closed high hat. There is limited room in the TR 606 plastic case, so I used a small five-conductor shielded cable to get the signals out of the case. Connect the shield to the ground and each of the five conductors to one of the TR 606 mix pots. It's best to connect to the "high side" of the pots so you'll always have full signals to your mixing console. Each of the conductors may be used as a separate signal out of each sound by connecting (conductor and ground) to any line input. Be sure to lower the levels on your mixing console before starting the TR 606. The original outputs can still be used normally. My TR 606 is great.

—Frank Stratton
Evensong Christian Music
Kensington, CA

More Help for the Bards

In reference to Ms. Kornhauser's of November 1982, on getting songs published, I would like to add to your response. There is a magazine called Songwriter Magazine, which deals with technical aspects of songwriting, getting songs published, etc. Its address is:

Songwriter Magazine
(Len Latimer Organization, Inc.)
6430 Sunset Blvd., Suite 908
Hollywood, California 90028

—John Donato
Connellsville, PA

Thank you for your help, Ms. Kornhauser and the other aspiring songwriters out there will be glad of the added source of information.

Will it Reach 360 in March?

Could you please tell me the address of 360 Systems? Your October issue carried an article on one of their keyboards, and it stirred my interest. I'd like to know how to reach 360 Systems.

—William A. Oliver
Princeton, KY

The address of 360 Systems is:

360 Systems
18730 Oxnard Street, No. 215
Tarzana, California 91356

In case you want to reach them by phone, their number is:

(213) 342-3127

Like, Have You Ever Heard Lavender Noise, Man?

Could you elaborate on the differences between pink and white and random noise? I know basically what they are and what the differences are between them, but I am having trouble with the limits of each definition. Help, please. I need some clarification.

—Ricky Coleridge
Philadelphia, PA

The Cameo Dictionary of Creative Audio Terms *bases its definition of white and pink noise on a definition of random noise. They explain that random noise is a mixture of all audio frequencies. This is perceived as a hiss. It can be due to irregularities in tape surface or small fluctuations in electronic component resistances—for example, thermal noise.*

White noise is random noise which contains equal energy for equal bandwidths. The 5,000 Hz bandwidth between 5,000 Hz and 10,000 Hz, or between 10,000 Hz and 15,000 Hz will have equal white noise energy. As you go up in frequency, the total energy per octave increases 6 dB/octave.

Pink noise is random noise modified to have an equal amount of energy in each octave. The 10,000 Hz to 20,000 Hz octave occupies a bandwidth of 10,000 Hz. The octave from 5,000 Hz to 10,000 Hz only occupies a 5,000 Hz bandwidth. But because there is a 6 dB/octave roll-off, both octaves have an equal amount of pink noise energy.

We hope this has been of some *help to you. And not just a bunch of printed noise.*

Hear today & Here tomorrow

Just two of the range of Europe's best selling sound reinforcement boards—now available in the United States.

SOUNDTRACS™
16-6 MONITOR

Possibly unique in specification, certainly in price. Incorporating all the necessary facilities for versatile 'on stage' monitoring. Can be linked to all '16 Series' boards by way of multi-pin connector and cable to provide a full 16 channel sound reinforcement mix.

- Phase reverse
- Master/master interlink
- Talkback to any of the six groups
- Balanced inputs/outputs
- 6 separate sends to masters
- Channel mutes and solos

Also available: 16-2, 16-4, 16-4-2, 24-4-2, 3 way crossover, 10 band stereo graphic.

SOUNDTRACS™
OMNI 16

Available either in a studio version (as shown) or as a mobile unit in flight case, the Omni 16 combines compactness with ease of operation and an exceptional track record of reliability. A number one choice for 16 channel sound reinforcement or 16 track recording at a very affordable price.

- Unique monitor/mix switching
- Balanced inputs/outputs plus unbalanced line
- Three Aux rails
- 100mm faders
- Line up 1K oscillator plus separate output

Full color brochure and technical specs from: SOUNDTRACS™

Soundtracs Inc. 262A Eastern Parkway, Farmingdale, NY 11735 (516) 249-3669

TALK BACK

Sunn Sheds Some Light

I own a Sunn Beta Bass amp. It's a 100 watt with 2 preamps integrated, and a remote switch to use one or both channels. It also has some wonderful thing which is called C-MOS overdrive and GMOS compression.

The problem I am experiencing is when using Channel A preamped to Channel B, I get great overdrive but when the compression is added, the signal becomes muddled. I want to try and get the colorful compressed sound with the overdrive.

Can you shed some light on this? I like the amp and I know there are capabilities I'm losing out on by my lack of understanding. You see, I am a musician whose knowledge of electronics is limited to playing electronic music. I am near desperate in need of some information. Even Sunn has been unable to help! There must be somebody out there who can help!

—Michael Frampton
Miami, FL

I spoke with Michael, and we determined that his problem was related to his use of effects units used in the accessory patch loops of his Beta Bass amplifier.

Effects units typically fall into two categories: pre-amp line level effects and guitar [low] level effects. Pre-amp line level effects typically are designed to work with signals up to 10 volts RMS [20 dBV]. An effect that is designed to be inserted between the guitar and amplifier input will usually be overloaded by signals above 1 volt RMS [0 dBV].

When the C-MOS compression drive control of a Beta is turned up, the level appearing at the pre-amp [accessory] output can be as high as 8 volts RMS [18 dBV]. For most line level effects, this is no problem, but for a guitar level effect, the input stage is being overdriven.

A quick survey of some typical effects devices shows that the maximum input level rating can be specified in dBV, dBm, or volts RMS. Now the problem is to figure out how to classify them, and compare them. I'll spare the mathematics details, and list the equivalent ratings in dBV, dBm and volts RMS.

I'll assume that any device that will overload at less than 1 volt RMS is meant to be used between the guitar and amplifier, not in an effects or accessory patch loop.

GUITAR LEVEL EFFECTS:
Typical maximum input level: 0 dBV, 1 volt RMS, 2.2 dBm or less

LINE LEVEL EFFECTS:
Typical maximum input level: 20 dBV, 10 volts RMS, 22 dBm.

Using these guidelines should help Michael and many other guitar players select and use their effects properly.

—Rex A. Baker
Electro-Acoustic Engineer
Sunn Musical Equipment Company
Tualatin, OR

The Shangri-La of Tascam

I have recently purchased a Tascam MM-20 mixer for use with a Teac A-3340-S four track recorder. The mixer gets the sound onto the tape without any problems, but there is one frustrating problem which as of yet I have been unable to solve. When I patch the A-B buss out into the monitor in, I pick up buzzes, hums and radio signals in the headphone circuit. I took the mixer into a Tascam repair service and, embarrassingly, found that the humming and buzzing did not happen in their building. I have now tried a variety of professional quality headphones and they all produce the same problem. When I plug the headphones into my stereo as the recorder, no such problem exists. The technicians at the service center say they have no idea what the problem could be. I have tried filtering my AC line through a line interference filter and also placed two .001 μf capacitors across the terminals of each channel to the ground lug of the headphone output jack. Both of the procedures were recommended by Teac when I called them up. The problem still exists and so does my frustration. Can you help.?

—Paul Taylor
Providence, RI

As you mention we have spoken already at Teac. This problem is not quite the same as the fellow who lives next door to a (Ham) radio operator. Commercial transmitters are many times more powerful than private radio rigs and can be expected to cause a host of problems in sensitive, high-gain audio systems. The solutions are pretty much the same however, and the degree of interference is proportional to your proximity to the transmitter, since commercial transmitters broadcast very clean signals and simply overwhelm the sensitive amplifying stages of gear like mixers and recorders.

As much as we would like to be able to operate anywhere, oblivious of things like radio transmitters, the laws of physics seldom yield, even to money, and there will be situations where things just don't work without taking extraordinary measures. Take for example the poor audiophiles of Boston and Seattle. In Seattle there is Queen Ann Hill in the center of the city—a natural place to locate all the city's transmitters for radio and TV, so that simple omnidirectional antennas reach all over town. Unfortunately, all over town there are users of audio gear as well as radios and TV's, and many have had to resort to servicing or modification or even moving to avoid the effects of transmitters too close. Manufacturers of audio gear can't tell you where to live, but they never recommend you move close to transmitters or other radio sources (powerlines can be a similar nuisance).

Generally, the design of electronic audio components such as mixers is done with a certain amount of radio interference in mind. Indeed, most audio designs are tested in Faraday rooms—special rooms covered by copper or copper screen, soldered together along the edges—to absolutely preclude any interference from outside, and controlled amounts of radio emission are fed into the area to determine the effects of radio interference on the design. Tascam mixers are designed this way, and the amount of radio interference rejected by mixers is determined to be sufficient to avoid interference effects in most areas where normal radio signal levels are present. Exactly what constitutes "normal" levels is arguable, but the number of complaints per year at Teac is less than 0.09% of the total sales of Tascam mixers, and without exception, every complaint comes from an area where within a few miles of a large transmitter—either television or commercial radio.

If you have tried all possible methods short of building a Faraday cage around your recording setup, and all have failed, you may have to consider one of several alternatives:

1) Change your gear. Move up to a more completely shielded unit.
2) Move your gear to another room or a basement where you can line the walls with chicken wire or screen as you would a Faraday shield.
3) Move away from the transmitter.

Thy Neighbor's Audacity

I own an 8-track recording studio (Tascam) which is in my home. My next door neighbor has a ham

Suddenly, everyone else has to start over.

Series Three from QSC. From this point on, every other power amplifier takes a step backward.

We've designed what will be the standard from which everything else is measured.

Series Three combines high-performance, reliability, and microscopic attention to detail, design and the execution thereof.

Our overall goal was to provide a series of reference-quality professional amplifiers designed specifically for major studios, touring companies and engineered sound installations.

We met our goal. As a matter of fact, we significantly exceeded it.

Take a look at the features of our Series Three amplifiers. High-efficiency circuitry, extended dynamic headroom, true dual-mono configuration, front-removable modular design. And we've packaged it all in a low-profile chassis. More power, less rack space.

We were so impressed with our prototype Series Three amps that we decided to take them into the field for numerous "A/B" listening comparisons. They were compared for audio quality and performance under a wide range of power requirement conditions. As we had expected, the response was overwhelmingly positive. The Series Three amplifiers stood a significant step above the others.

The moral of the story: Why settle for a product that's only outstanding in a few areas? QSC Series Three is a comprehensive design approach that combines exceptional audio performance, solid reliability, state-of-the-art features, and more power in less rack space.

So we urge you to look into our Series Three amplifiers. Because while everybody else is looking where they've been, QSC is stepping into the future.

QSC Audio Products. 1926 Placentia Avenue, Costa Mesa, CA 92627. Phone: (714) 645-2540.

QSC AUDIO

ALABAMA
Big Bear Audio
Sheffield

CALIFORNIA
All Star Sound
Ventura

J L Music
Montclair

Nadines
Hollywood

Whittier Music
Whittier

CONNECTICUT
Altel Sound
Bridgeport

La Salle Music
W. Hartford

FLORIDA
Modern Music
Ft. Lauderdale

Mr. Hank, Inc.
N. Miami

GEORGIA
Music Mart
Smyrna

Oap Audio
Buford

Rock 'N Road Audio
Atlanta

ILLINOIS
Beardsley Light & Sound
Moline

D.J.'s Pro Audio
Cicero

KANSAS
Superior Sound
Wichita & Kansas City

MARYLAND
Washington Music Center
Wheaton

MASSACHUSETTS
Berkshire Prof. Sound
Dalton

Sid Stone
Boston

MICHIGAN
Cook Christian Supplies
Alma

Guitar & Drum Center
St. Clair Shores

John Kik & Assoc., Music
Grand Rapids

MINNESOTA
MSC Pro Sound
St. Paul

NEW JERSEY
Caiazzo Music
Freehold

NEW YORK
Alex Musical
New York City

Only Guitar Shop
Clifton Park

Universal Sound
Lake Grove

Whirlwind Audio
Rochester

NORTH CAROLINA
Associated Sound Prod.
Raleigh

Duncan Pro Sound
Winston-Salem

PENNSYLVANIA
Ro'Cel Electronic
Pittsburgh

RHODE ISLAND
Rhode Island Musical
Pawtucket

SOUTH CAROLINA
Gordon Light & Sound
Greenville

TENNESSEE
Keiths Corner Music
Cookeville

Lowrance Sound Co.
Union City

Spectrum Sound
Nashville

TEXAS
Crossroads Audio
Dallas

L.D. Systems
Houston

Recorder Center
Dallas

VIRGINIA
Stage Sound
Roanoke

WEST VIRGINIA
The Pied Piper
Huntington

WISCONSIN
Regenberg Music
Middleton

Uncle Bob's Music
Milwaukee

CIRCLE 18 ON READER SERVICE CARD

radio, and whenever he broadcasts, it totally interferes with any recording I'm trying to do. I keep getting loud call letters and static. There's no reasoning with him, as he's very stubborn. Is there anything I can do to filter out this interference?

—Jon Lester
Burton, OH

I called the American Radio Relay League (ARRL) in Washington, D.C., and talked at some length to their legal counsel. Here's what I found.

Most Radio Amateurs belong to the ARRL. Radio Amateurs call themselves HAMS and set themselves apart from the majority of CB users for a number of reasons, mostly having to do with the difficulty of obtaining radio operating licenses and all the expertise needed to build up and run a Ham rig, which is considerable. The ARRL has guidelines for its members to help them stay on the good side of the public and the FCC, and these guidelines include operating transmitters in good condition, tuned to minimize spurious signals and radio emissions other than those intended for broadcast.

There are very few Hams compared to the legions of CB'ers who run amok on the air with precious little in the way of checks and balances to curtail illegal language and over-power transmissions not to mention CB sets that never get technical attention until they break down—often the source of spurious signals unknown to their owners.

ARRL informs me that uncooperative hams are very rare, since one of the most hallowed codes of radio amateurs is public service and national assistance in case of emergency or disaster. If your radio-nemesis turns out to be a Ham rather than a CB'er, you are asked to contact the ARRL ([202] 296-9100) so that they might know of the situation and possibly mediate some sort of agreement between you and your neighbor. In severe cases, the FCC might be called in and "quiet hours" might be assigned to the Ham rig so that your business may be operated free of interference. I might add, that if the radio rig in question is running properly but simply overpowering your sensitive audio electronics by virtue of available antenna power, you are as much responsible for a possible electronic fix to your gear as is the Ham for keeping his transmitted signal clean and free of spurious signal components.

If the radio rig turns out to be CB, your only recourse is to call the FCC field office and ask to have the transmissions monitored for legality, and if the signal is determined to be illegal either because of spurious signal components or power beyond the legal limits for CB, the FCC will help out by shutting down the offending transmitter if warnings are not heeded.

One thing to keep in mind is that there is no clear remedy to situations like yours, and that solutions often take a long time, even years, if it becomes necessary to take court action against obnoxious radio sources, especially if they are "legal" in the sense of operating properly within FCC and ARRL guidelines.

Concerning filtering the radio out of your recording equipment, there are many ways to reduce or eliminate radio from audio lines. Here is a list of measures that may be used, listed in order of difficulty, with their effectiveness noted on a scale of one to ten next to each:

- Use of braided-shield cable of high quality with good braid coverage and good insulation integrity. (3 to 10).

- Avoidance of ground loops that can carry current and radio signals through chassis parts and cabling. (7 to 10). (Eliminating ground loops is potentially very tricky.)

- Connecting small capacitors across audio inputs (2 to 5).

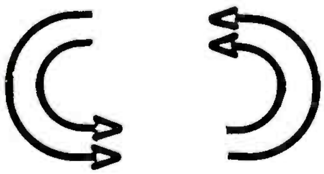
- Using small ferrite bead inductors on the input wires of audio gear (1 to 3).

• Use of A.C. line interference filters to power cords of audio gear or to the power supplies of each piece in the audio system.* (2 to 6).

• Use of copper bar or pipe grounding system to all chassis and elimination of all other ground connections in the audio system. (4 to 8). (This can be real tricky and is to be avoided if you are not familiar with ground loops.)

• Placing all audio equipment in grounded enclosed rack cabinets. (5 to 10). (Very expensive.)

*A.C. line interference filters are available from many sources. Some of these include:



J. W. Miller Division of
Bell Industries
19070 Reyes Avenue
P.O. Box 5825
Compton, CA 90224 (213) 537-5200

SGL Waber Electric
Department W
300 Harvard Avenue
Westville, NJ 08093 (609) 456-5400

MCG Electronics Inc.
160 Brook Avenue
Deer Park, NY 11729
(516) 586-5125

Corcom Inc.
1600 Winchester Road
Libertyville, IL 60048
(312) 680-7400

Control Concepts Corp.
333 Front Street
Binghamton, NY 13905
(607) 724-2484

Electronic Specialists, Inc.
171 South Main Street, Dept. PE
Natick, MA 01760 (617) 655-1532

—Drew Daniels
Applications Engineer
Tascam Production Products
Montebello, CA

Recording Techniques Part 11

by Bruce Bartlett

One of the most exciting moments in recording comes when the finished mix is played over the big studio monitor speakers. The sound is so clear you can hear every detail, and so powerful you can feel the deep bass throbbing in your chest.

The monitor system is used to listen to the output signals of the console of the tape recorders. It consists of the console monitor mixer, crossovers, the power amplifiers, loudspeakers, and the listening room. The power amplifiers boost the electrical power of the console signal to a sufficient level to drive the speakers; the speakers convert the electrical signal into sound, and the listening-room acoustics affect the sound from the speakers.

This article will cover crossovers, speakers, and room acoustics. Other parts of the monitor chain, as well as monitor installation and usage, will be explored in the next installment.

The monitor system is a critical link in the recording-and-reproduction chain, for it provides the feedback that tells what you're doing to the recorded signal. You adjust the mix according to what you hear through the monitors; you also use the monitors to judge the effectiveness of microphone techniques. Clearly, the monitor system affects the settings of many controls on the console, as well as the microphone selection and placement. And all those settings affect the sound you're putting on tape. So, using inadequate monitors can result in a poor-sounding product coming out of your studio.

For example, if your monitor speakers are weak in the bass, you will tend to boost the bass in the mix until it sounds right over those monitors. But when that mix is played over speakers with a flatter response, it will sound too bassy—because you boosted the bass in the control room. So, using monitors with weak bass results in bassy recordings; using monitors with exaggerated treble results in dull recordings, and so on. In general, colorations in the monitors will be inverted in the final tape. That's why it's so important to use an *accurate* monitor system—one with a wide, smooth frequency response. Such a system lets you hear exactly what was recorded.

Monitors also have another function: to let you hear the program as the listener will hear it. Since the purpose of recording is to please this customer, you must tailor the mix to sound pleasant over the kind of speakers the listener will have. Remember, when you're doing a mix over a particular monitor system, you're creating what the home listener will hear, assuming he or she has speakers and a listening room like yours.

Suppose all recording engineers and home listeners had the same speakers, the same speaker placement, and the same listening-room acoustics. Then, engineers could rest assured that the sound they created over their monitors would be duplicated in everyone's home; every home listener would hear what the engineer and producer *intended* them to hear.

But everyone has different speakers, speaker placements, and listening rooms. A recording will sound different on every system it is played on. So, to please as many listeners as possible, you need to use a monitor system that sounds like a typical home system used by the intended audience.

Thus we have a need for two kinds of monitors: (1) *accurate*—to hear what's really there, and (2) *typical* (possibly narrowband)—to simulate a home listener's system.

Room Acoustics

The first step in ensuring an accurate monitor system is to work on the control-room acoustics. You may want to review Part 2 of this series (on room acoustics) in the March 1982 issue of *MR&M*. Let's start by explaining how room acoustics affect the monitored sound.

Sound waves leaving the speaker strike the room surfaces. At those surfaces some frequencies are absorbed, while other frequencies are reflected back to the listener's ears. The sound waves reflected from the room surfaces blend with the direct sound from the speakers to make the composite sound that is perceived. Only those reflections arriving up to 20 to 65 milliseconds after the direct sound will blend or fuse with the direct sound to affect the perceived spectrum.

Suppose the walls are covered with carpet so that they absorb only the high frequencies. Then the walls will reflect mainly the low frequencies. When you listen to a speaker playing in such a room you will hear the direct sound from the speaker, plus the bassy wall reflections, giving a total sound that is bass-heavy. Now suppose the walls are made of wood paneling mounted on studs. Such a vibrating surface absorbs lows and reflects highs. The total sound you hear probably will be thin and overly bright. Clearly, the room surface should reflect (or absorb) all frequencies about equally to avoid coloring the sound of the speakers. Equal absorption (+/-25 percent) from about 250 Hz to 4000 Hz is usually adequate. As described in Part 2, you can use flexible panels to absorb lows, in combination with fibrous materials or foam to absorb highs. Or use thick fibrous material spaced out from the wall and ceiling.

There's an alternative room treatment that takes up a lot of space but easily achieves equal absorption at all frequencies. You cover all the walls and ceilings with about 3 feet of fiberglass. This treatment absorbs most frequencies equally. Then you add 1-inch thick plywood panels, one at a time, until the desired reverberation time is reached. The thick panels reflect all frequencies equally. Normally, these panels are spread evenly around the room (as in *Figure 1*). Alternatively, you can start with a room that has solid walls and add patches of 3-foot thick fiberglass until the desired reverb time is achieved.

Room resonances or standing waves can cause some notes to blare out and cause other notes to disappear. To minimize the audible effect of standing waves, aim for an even distribution of room resonance frequencies. This can be achieved by using or building a control room with dimensions that are not multiples of each other. Some preferred ratios of

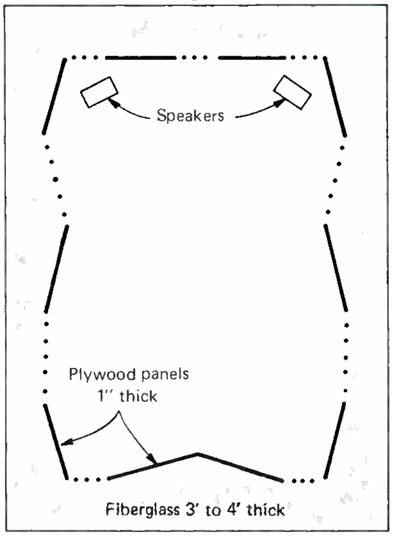


Fig. 1. Acoustical treatment providing equal absorption vs. frequency.

height, width, and length are given below:

1:1.14:1.39
1:1.17:1.47
1:1.26:1.41
1:1.28:1.54
1:1.45:2.1

1:1.47:1.7
1:1.6 :2.33
1:1.62:2.62

Taking the top ratio as an example, if the ceiling height is 10 feet, the room width should be 11.4 feet and the length should be 13.9 feet for best distribution of resonances.

Room acoustics also affect the decay-in-time of the sound coming from the speakers. When a note in a reproduced program suddenly ends, the sound of that note continues to bounce around the room, causing echoes and reverberation that prolong the sound. This long decay of sound is not part of the program. So the control room should be relatively dead—that is, it should have a short reverberation time. A typical living room has a reverberation time of about 0.4 second; the control room should too, so the engineer will hear about the same amount of ambience that a home listener will hear. A totally dead room is uncomfortable to listen in.

To absorb reflections, use sufficient sound-absorbent materials on the walls, ceiling, and floor. Flutter echoes can be eliminated by making the walls non-parallel.

Reducing room reflections also improves the stereo imaging of the monitors. All the information about sound-image location is in the direct sound coming from the speakers; wall reflections can only confuse the listener as to the correct location of the sound images.

The control room should be acoustically isolated from the studio so that you hear only the sound from the monitors, not the live sound from the studio musicians. In a home studio, you can achieve isolation simply by putting the control-room equipment in a room far removed from the studio, with the doors closed. It also helps to monitor with closed-cup headphones. To build a control room with good isolation near the studio, you may need to use double-wall construction with staggered studs and fiberglass insulation between the two walls. The door between the control room and studio should be solid wood and should be weather-stripped all around—including underneath. Use a double-pane window (mounted in rubber) between the control room and studio.

More information about control-room construction can be found in *How to Build a Small Budget Recording Studio from Scratch* by F. Alton Everest, Tab Books, Blue Ridge Summit, PA, 17214, 1973.

Speakers and Crossovers

Most studio monitor loudspeakers are designed for high efficiency, typically using a ported 15-inch woofer for the lows and a horn-loaded compression driver for the highs. The bass-reflex port (or a passive-radiator cone) resonates at low frequencies to reinforce the bass; the horn efficiently couples the compression driver's sound to the air, like a megaphone.

One-way speaker systems use a single-cone speaker. An example would be a mini-monitor used to simulate compact stereos and car radios. Two-way systems (the most popular) have a woofer and a tweeter. Three-way speakers use a woofer, a mid-range driver (horn-loaded or cone), and a tweeter. Four-way systems cover the widest range by adding a super-tweeter.

A *crossover* or *dividing network* is a circuit that divides the monitor signal into two or more frequency bands. In the crossover, a low-pass filter sends the low frequencies to the woofer; a high-pass filter sends the high frequencies to the tweeter and keeps out low-frequency components that could damage the tweeter. Some crossovers include a band-pass filter to route middle frequencies to a mid-range driver.

The *crossover frequency* is the frequency above which the woofer stops working and the tweeter takes over. The low-pass filter passes frequencies below the crossover frequency; the high-pass filter passes frequencies above the crossover frequency. This frequency is set below the upper limit of the woofer and above the lower limit of the tweeter—typically 500 to 2500 Hz. There may be an additional crossover around 5000 Hz between a mid-range driver and a tweeter.

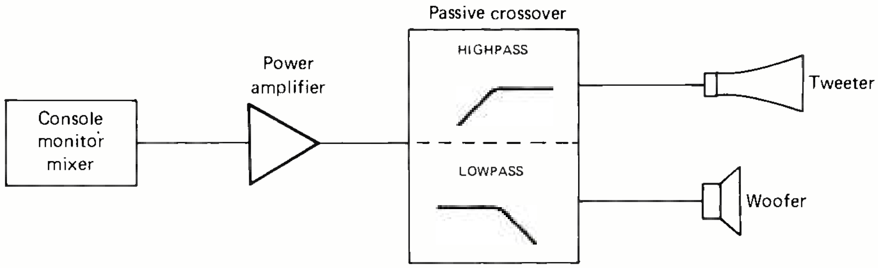


Fig. 2. Monitor system using a passive crossover.

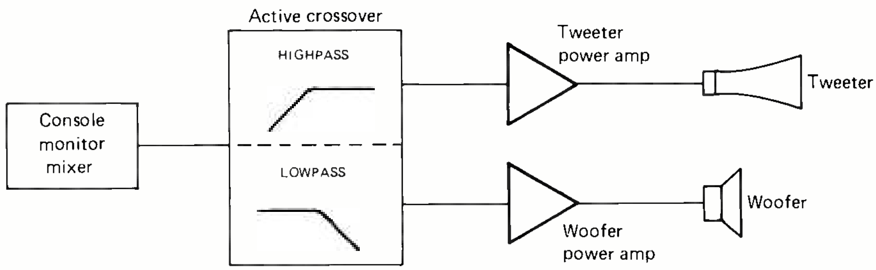


Fig. 3. Biamped monitor system using an active crossover.

If the crossover is connected after the power amplifier, the crossover operates at high power levels and is called a passive crossover *(Figure 2)*. If the crossover precedes the power amps, it operates at line level and is called an active crossover or electronic dividing network *(Figure 3)*.

Active crossovers are used in biamped systems, in which the woofer and tweeter are driven by separate power amplifiers. The active crossover is connected ahead of the power amps. Low-frequency signals from the crossover go to the power amp driving the woofer; high-frequency signals from the crossover go to the power amp driving the tweeter. Because tweeters are more efficient than woofers, the tweeter amp can be about ¼ the power of the woofer amp.

Biamping is claimed to have several advantages:

• Distortion components caused by clipping the woofer power amplifier will not reach the tweeter, so there is less likelihood of tweeter burnout of the amplifier clips. In addition, clipping distortion in the woofer amplifier is made less audible.
• Intermodulation distortion at high levels is reduced.
• Peak power output is greater than that of a single amplifier of equivalent power.
• Direct coupling of amplifiers to speakers improves transient response—especially at low frequencies where damping factor is improved.
• Biamping reduces inductive and capacitive loading of the power amplifier.
• The full power of the tweeter amp is available regardless of the power required by the woofer amp.

Three-way speaker systems can be triamped, or can be biamped by putting a passive crossover before the tweeter *(Figure 4)*.

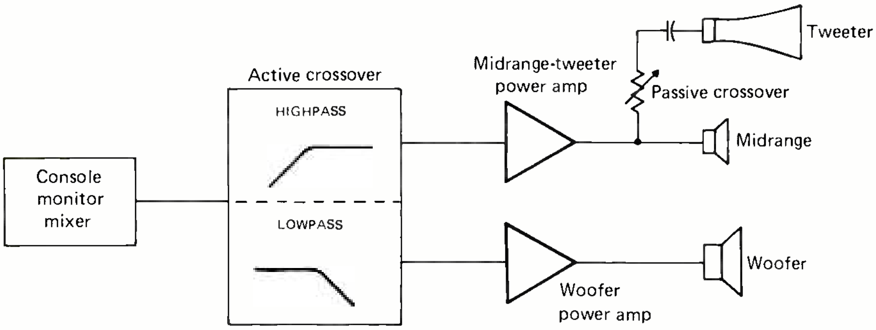


Fig. 4. Biamped monitor system using an active crossover and a passive crossover.

Speaker Requirements

The requirements for an accurate studio monitor are as follows:

Wide, smooth frequency response. This should be from at least 40 Hz to 15 kHz, ±4 dB or less, to ensure accurate tonal reproduction.

Controlled dispersion. A studio monitor should focus its sound on the listener and prevent radiation to the sides and rear of the speaker. That's to reduce reflections from nearby surfaces which can degrade the speaker's frequency response and stereo imaging. In monitoring a recording, we want to hear what's on tape rather than hearing sound reflections from the room surfaces.

Thus, a monitor should have controlled dispersion, with a relatively narrow spread of sound leaving the loudspeaker. Dispersion is measured in *coverage angle*, the off-axis angle at which the speaker output is down 6 dB compared to the on-axis output. The horizontal coverage angle of a monitor should be about 60 to 90 degrees—just wide enough to cover the console area evenly. Ideally, the angle should be maintained at all frequencies, so that people seated anywhere behind the console will hear the same tonal balance.

Good transient response. This is the ability of the speaker to accurately follow the attack and decay of musical sounds. Transient response is aided by aligning the acoustic centers of the woofer and tweeter so that their signals arrive at the listener at the same time.

Clarity and detail. The listener should be able to hear small differences in the sonic character of instruments, and to sort them out in a complex musical passage.

Low distortion. This is a necessity for low listening fatigue.

High efficiency. Efficiency is the ratio of sound power output to electrical power input. A high-efficiency speaker is louder than a low-efficiency speaker when both are driven by the same input power. High efficiency means less overheating because less power is needed for the same loudness.

High sensitivity. Sensitivity is the sound pressure level a speaker produces at 4 feet when driven with 1 watt of pink noise. This noise is either band-limited to the range of the speaker or is a 1/3-octave band centered at 1 kHz. A sensitivity specification of 93 dBSPL is considered high (typical of studio monitors); a specification of 85 dBSPL is considered low (typical of home bookshelf speakers).

High output capability. This is the ability of a speaker to play loud without burning out. It's often necessary to monitor at high levels to hear quiet details in the music.

The power rating is divided by 10 to allow for 10 dB peaks in the music, and 6 dB is subtracted to allow for the inverse-square-law loss in level when the distance from the speaker is doubled (from 4 feet to 8 feet).

More volume can be obtained by wiring an identical pair of speakers in parallel; that divides the power equally between them. Then you can double the amplifier power, which boosts the level 3 dB. Stacking the speakers vertically will result in an additional 2 dB at low frequencies due to coupling between drivers, plus an additional 3 dB due to increased directivity.

Plus, when you record musicians who play loudly in the studio, it can be a letdown for them to hear a quiet playback. Consequently, a maximum output capability of 110 dBSPL is typically required.

To calculate how loud a given speaker can play, use the following formula as a rule of thumb:

$$dBSPL = 10 \log P/10 + S - 6\,dB$$

where dBSPL = the peak sound pressure level obtain-able at 8 feet P = the continuous sine-wave power rating of the speaker and S = the sensitivity rating in dBSPL/ 1 watt/4 feet

Current Speakers In Use

Regarding large monitor speakers, the latest models of the major manufacturers are the UREI A series, the JBL 4400 series, and the Electro-Voice Sentry 500. Other well-known monitors are made by Westlake, Hidley, and Tannoy.

Some popular earlier models are the UREI 813TA, UREI 815TA, Altec 604 with the Mastering Lab Crossover, JBL 4300 Series, Audio-techniques Big Red and Super Red, and Electro-Voice Sentry 4.

Some popular mid-sized monitors are the JBL 4311B, JBL 4313, and the Dahlquist DQM-9. Models by other manufacturers such as ADS and B & W are also well worth auditioning.

If you operate a small home studio, you may just want to use your hi-fi system for monitoring. Some speakers and headphones I've found that provide reasonable accuracy for a low price are the EPI 100 speakers and the Yamaha YH Series headphones. That is not necessarily a recommendation; use your own judgement.

Next month we'll continue our discussion of monitoring, covering such topics as speaker placement, power requirements, room equalization, monitor usage, headphones, and the cue system. Stay tuned!

THE PRODUCT SCENE

AUDIO TECHNOLOGY'S LED DISPLAY

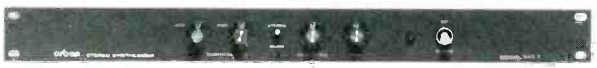
Audio Technology announces the first user programmable digital peak hold LED audio level display—the Model 511. The Model 511 incorporates highly linear, dual-phase peak detectors (pat. pend.), and both line level (dBm) and power level (dBw) inputs.

In use, the highest LED is digitally held while the lower LEDs continue to function in real time. The peak hold circuitry functions in both the dBm and dBw modes and is operated by turning the mode selector button on the front panel to the desired hold time.

The 511 will find application in studio and broadcast consoles, disc mastering, tape duplication, audio-visual, communication and sound reinforcement. An entire tape can be monitored and the highest level stored prior to being duplicated or cut on a disc, sync pulse level can be measured and stored for comparison, amplifier output can be stored—all without an operator present to record levels. Continuous level monitoring is also enhanced with shorter hold times. The Model 511 is available in horizontal or vertical format. Rack panels are available for combinations of one, two, or eight displays. The suggested list price is $219.95.

CIRCLE 37 ON READER SERVICE CARD

ORBAN'S STEREO SYNTHESIZER



Orban Associates, Inc. of San Francisco, California announces the availability of the new 245F Stereo Synthesizer. The 245F is an improvement of the popular 245E version, adding balanced input, output transformer option, RF filtering on the audio inputs and outputs, and AC line filtering.

Orban's patented stereo synthesis technique allows any mono source to be converted to pseudo-stereo with no phase cancellation in the mono original. Mono material such as single tracks in a studio, DJ mikes, mono cart machines, old records, spots and promos, and TV audio can be converted into pseudo-stereo to create a spatial effect. The price of the Orban 245F is $399.00.

CIRCLE 38 ON READER SERVICE CARD

NEW STEREO COMPRESSOR/LIMITER

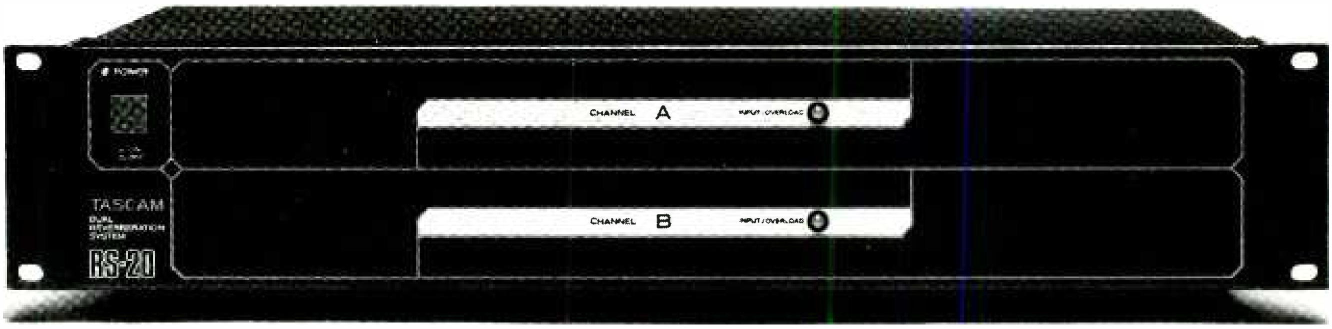
Modular Audio Products introduces the Model 7102 Stereo Compressor/Limiter. The 7102 contains two VCA units controlled by a single DC source derived from a circuit using a true RMS to DC converter. The input is bridging unbalanced, the output is low impedance unbalanced. The dynamic gain reduction characteristic of the 7102 is a "soft-knee," gradually increasing the rate of gain reduction as more compression is required. Front panel controls are: input (threshold) level, compression ratio control, output level control and a slow/fast release time switch. Two LEDs show normal and clipping levels. Left and right VU meters can be switched to input or compressed output for visual monitoring. A power on/off circuit breaker completes the front panel controls. The 7102 comes with an integral power supply and barrier type terminal strips for audio input and output connections. The unit is housed in a compact 1¾-in. (one rack unit) high, 9-in. deep, fully enclosed case.

CIRCLE 40 ON READER SERVICE CARD

PARAMETRIC EQ FROM TASCAM

One of the new devices in Tascam's "System Enhancement Series" is the PE-40, a 4-channel, 4-band fully parametric equalizer. Each channel has four sets of continuously variable Q, gain and frequency controls. The four bands overlap for greater control: 40-800 Hz, 200-4,000 Hz, 500-10,000 Hz, and 800-16,000 Hz. Two or more of the PE-40's channels can be cascaded when maximum control is needed, for example, in precisely "tuning out" multiple room modes. To reduce rumble, wind noise and feedback, each channel has switchable 60 Hz, 18 dB/octave and 160 Hz, 6 dB/octave high pass filters, as well as a 15 kHz, 12 dB/octave low pass filter to reduce hiss.

CIRCLE 41 ON READER SERVICE CARD



TWEAK FREE REVERB. REVERB. REVERB.

Many reverbs come with level controls, but not the RS-20—it's tweak free. After all, your mixer has echo send and return controls, so why pay twice for the same thing. Just set the rear panel sensitivity switch, and a pair of bi-color LEDs on the front panel help you set the mixer's send level. That's it.

Limiters ahead of the RS-20 drive amps then prevent overdrive or "twangy" spring sounds caused by high energy transients inherent in plucked guitar strings, etc., so the RS-20 remains "squeak free."

This exciting new unit incorporates a proprietary design with three different sized springs on each channel. Here's why: Most conventional single-spring reverbs have poor high frequency response, and those that don't usually compensate by "shunting" high frequencies directly through their reverb with a capacitor, a short-cut that cheats you of high frequency delay. The RS-20's multiple springs extend the frequency range at least an octave above conventional units (without a shunt). Also, three springs let us scale the decay time to the frequency for superbly natural sounding reverb.

You've got to hear it—sound that rivals high quality plates and digital systems, for a whole lot less. Even at twice the price, nothing beats the RS-20 Dual Reverb.

For additional information, see your TASCAM dealer, or write TASCAM Products, 7733 Telegraph Road, Montebello, CA 90640, (213) 726-0303.

TASCAM
TEAC Production Products

DESIGNED BY POPULAR COMPLAINT.

CIRCLE 22 ON READER SERVICE CARD

NEW TUSC AMPLIFIER

Tusc Amplification has recently added several models to its "Prestige Series" of tube amplifiers. The DF 55 Porta-Bass, designed for bass and keyboard, is a compact unit featuring one heavy duty, 15-in. Fane bass speaker and flip-top power head for compactness. The unit has complete programmability, enabling the user to change from one setting to another by a footswitch.

CIRCLE 42 ON READER SERVICE CARD

SHURE'S 586 SERIES

Shure Brothers of Evanston, Illinois has announced the introduction of the Shure 586 Series of Cardioid (unidirectional) Dynamic microphones. The 586 Series is an improved version of Shure's 585 Series. It is designed to meet the current demand for fast and easy microphone-cable connection, incorporating a standard 3-pin professional connector which replaces the 585 series' screw-on Amphenol-type connector.

The 586 Series also incorporates a new, internal anti-pop screen and a shock mount that reduces undesirable wind and handling noise. Other features include an extended low-end frequency response, a lockable on/off switch, a feedback-reducing cardioid pickup pattern, and a die-cast handle. Each microphone in the series is packaged with a swivel adaptor for quick mounting and stand adjustment.

The Shure 586 is available in four versions: 586SA-LC (high impedance, supplied without cable), 586SB-LC (low-impedance, supplied without cable), 586SA-C (high impedance, supplied with 6.1 m [20 ft.] cable with ¼-inch phone plug on the equipment end), and 586SB-CN (low impedance, supplied with 6.1 m [20-ft.] cable with 3-pin XLR connectors on both ends). User net prices are: $91.50 for the 586SA-LC and 586SB-LC, $106.00 for the 586SA-C, and $112.50 for the 586SB-CN.

CIRCLE 43 ON READER SERVICE CARD

A DRAGON NO PRINCE WOULD SLAY

Nakamichi U.S.A. Corporation announces the Dragon, a new auto-reverse cassette deck featuring Nakamichi Auto Azimuth Correction (NAAC). Unlike manual and automatic azimuth correction systems that require special signals to operate, NAAC determines optimum azimuth from the program material itself.

The NAAC system employs a specially created playback head that derives two signals from the same track. A precision phase comparator determines azimuth misalignment by inspecting these signals in a specific band of frequencies and adjusts the playback head accordingly. NAAC does not require that left and right channels contain similar information. Thus, full stereo operation is ensured.

Dragon includes discrete three-head technology, individual left- and right-channel calibration of bias and sensitivity for three tape types, microprocessor transport control, vibration-absorbing chassis, Dolby B- and C-type noise reduction, 2-speed Easy Cue, 2-speed Auto Fade, wide range peak-responding electronic meters, punch-in recording, and remote control capability. In addition to defeatable MPX filters for FM recording, Dragon has switchable subsonic filters that prevent tape overload and intermodulation distortion when recording a warped phono disc. And, a new Auto Rec Pause feature automatically switches from Record to Record Pause whenever a program break of more than 10 seconds is detected. The Dragon has a retail price of $1,850.

CIRCLE 44 ON READER SERVICE CARD

ELECTRONIC SPECIALISTS CONTROL AUDIO POLLUTION

Electrical pollution can drive audio systems bananas! Power line electrical noise, hash and spikes can often cause erratic component operation. In addition, severe spikes from lightning or heavy machinery may damage expensive hardware. Many systems create their own pollution. Power amp/preamp interactions often create enough electrical interference to degrade an entire program. Nearby electronic equipment may also be affected.

Electronic Specialists of Natick, Mass. recently announced the development of the Magnum Isolator. It is designed to control severe electrical pollution. Incorporating heavy spike/surge suppression, the Magnum Isolator features four individually quad-Pi-filtered AC sockets. Equipment interactions are eliminated and disruptive/damaging power line pollution is controlled. The Magnum Isolator will control pollution for an 1875 watt load. Each socket can handle a 1000 watt load. Severe AC power line audio pollution can be controlled with the model ISO-17 Magnum Isolator, priced at $181.95.

CIRCLE 45 ON READER SERVICE CARD

6 Ch CX601 Mono $599 with 200 watt amp (list $1195)
2 Ch CX1201 Mono $799. Add $200 for 300W amp (list $1595)

Ch SX602 Stereo $649. Add $150 for 200W stereo amp (list $1295)
Ch SX1202 Stereo $999. Add $250 for 300W stereo amp (list $1995)

12 Ch MX1202 Deluxe Stereo 12x2x1 $1195 (list $2595)
16 Ch MX1602 Deluxe Stereo 16x2x1 $1495 (list $3395)
16 Ch MX1608 Deluxe 8 Out 16x8x2 $2595 (list $7500)

Only the Best Will Do!

Frank Zappa spared no expense in setting up a mobile recording studio for his recent tour. Along with some of the most exotic audio gear available he is using three CARVIN MX series mixers.

We at CARVIN have a personal commitment to making only the finest professional boards. We have taken our time to develop the "State of the Art" audio circuitry found in every CARVIN mixer so that you can enjoy the same transparent sound that Frank Zappa and other CARVIN owners enjoy.

Only the finest components are incorporated in our products: high slew rate-low noise integrated circuits, long life potentiometers, velvet smooth faders, military type wiring harness, professional VU meters, heavy-duty switches and connectors, and solid oak wood. Special manufacturing and testing techniques give the CARVIN boards fail-safe reliability whether on the road or in the studio.

We sell "DIRECT ONLY" from our manufacturing plant in California. That means big savings because there are no middle men. At CARVIN we have been standing behind our product since 1946 and we're dedicated to making the very best! We give fast Mail-Order service with low freight rates! When in southern California visit our show rooms for a personal demonstration.

Another big plus at CARVIN is our 10 day trial with your money back if you're not satisfied. Check our super low prices on over 25 different Carvin sound systems, 800 watt power amps, and a full line of guitars and amplifiers. Master Card/VISA accepted. Send $1 for your 80 page Color Catalog. Send $2 to RUSH by 1st Class mail.

Available Direct Only!

TOLL-FREE 800-854-2235
Calif. 800-542-6070

CARVIN

Dept. MR-46, 1155 Industrial Ave., Escondido, CA 92025

CIRCLE 23 ON READER SERVICE CARD

CARVIN CATALOG
☐ $1 for Bulk Mail
☐ $2 for First Class Mail
Name
Street
City
State
Zip
MR-46

IBANEZ INTRODUCES NEW DIGITAL DELAY

The DM1000 is based on the latest microprocessing digital technology in digital circuitry. It is enclosed in a 19-in. steel casing. The new approach encoding-decoding system and digital processor provide time delay ranges of 1.75 msec to 900 msec with full bandwidth for all delay ranges.

The DM1000 input is useful to a variety of instruments and sound reinforcement applications. In addition, the three types of output—dry, mix, and inverted mix—facilitate the various stereo operations. The DM1000 features a footswitchable infinite hold (repeat) and a ±12 dB equalization tone control at 6 kHz. The suggested retail price for the DM1000 is $449.00. CIRCLE 46 ON READER SERVICE CARD

COAXIAL LOUDSPEAKERS

Professional Audio Systems has recently introduced a complete line of coaxial loudspeakers for musicians who require accurate reproduction of complex transient sounds. All loudspeakers may be ordered with a Time Offset Correction network that eliminates phase wash/time smear.

Loudspeakers are available in cabinet form and as raw frame components. The 12-in. and 15-in. coaxial components will accept most 1-inch compression drivers. Also available is a Passive Crossover Network with built-in Time Offset Correction which can be used with various 1-inch compression drivers. "Monster Monitor" stage monitor will also eliminate phase wash when used with the Time Offset Correction network.
CIRCLE 47 ON READER SERVICE CARD

FOSTEX'S COMPRESSOR/LIMITER

Fostex Corporation of America introduces a new 2-channel gated compressor limiter, Model 3070. Gain reduction is accomplished via a VCA circuit which is controlled by pulse width modulation. In the VCA circuit there is an electronic switch which opens and closes at a rate of 200 kHz. By varying the length of time this electronic switch is open or closed during each cycle, signal energy is reduced without distorting the program.

Compression ratios are continuously variable from 1:1 through infinity:1. Attack times are continuously variable from 0.2 msec. through 20 msec., and release times are continuously variable from 50 msec. through 2 msec.

Other features include a noise gate function with independent threshold setting, LED displays indicating gain reduction in calibrated decibel readings, and dual mono or linked stereo modes. Maximum gain reduction is 32 dB. The suggested list price of the new Fostex 2-Channel Gated Compressor/Limiter is $400.00. CIRCLE 48 ON READER SERVICE CARD

TEAC'S Z-7000 CASSETTE TAPE

Teac's new Z-series of cassette decks was introduced at the Consumer Electronics Show in Las Vegas. The top of the line is the Z-7000, with fully automatic bias/level/EQ calibration and Teac's Auto Tape Selector, which automatically sets bias and EQ parameters according to the type of tape used. It also features several exclusive computer automated locating functions, such as Search to Zero; Search to Cue; Search to Record and Intro Check, which plays the first ten seconds of each selection in sequence. Other features include three independent high-precision motors; a dbx disc button; Teac's Spot Erase System, for editing out program material after it's been recorded; automatic fade-in/fade-out; a built-in headphone amplifier with volume adjust; pitch control; power eject and MOL balance controls. Suggested retail price is $1,800.00.
CIRCLE 49 ON READER SERVICE CARD

Full Bandwidth Digital Delay... Naturally!

The ***EFFECTRON***™ series is the *only* low cost, full bandwidth, high performance line of digital delay processors. A real technological breakthrough unmatched by anyone!!!

At last—you can afford high quality, natural sounding digital delay.

Visit your local dealer and check it out!

Listen and compare . . .

ADM-256
¼ Second Delay
$499*

ADM-64
Flanger/Doubler
$399*

ADM-1024
Full Second Delay
$699*

Music & Electronics...Naturally! DeltaLab

*Manufacturer's Suggested Retail Price

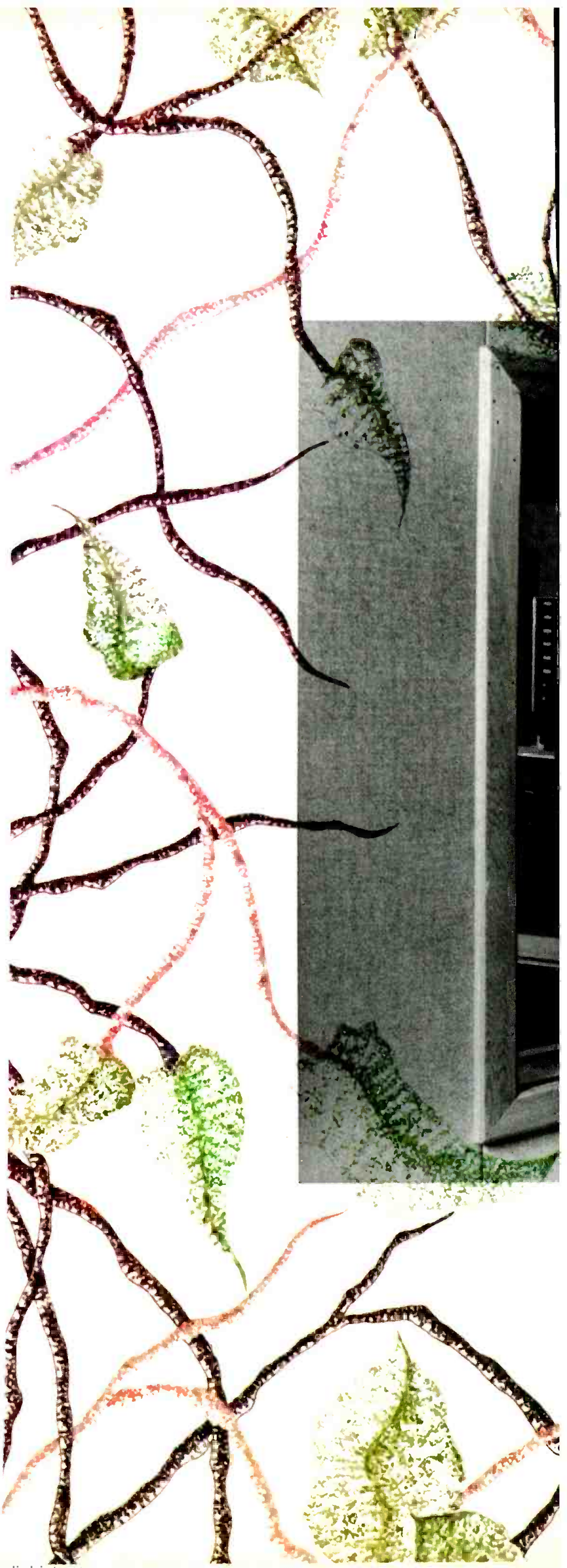
Spyro Gyra

by amy gordon

Spyro Gyra, led by Jay Beckenstein and Rich Calandra, have made continuous progress since their music began to take hold in upstate New York in the late seventies. With the release of their fifth album, *Incognito*, they have announced the opening of their new studio, Bear Tracks, in which their latest record was mixed.

The SSL 5000 computer-automated board and the George Augspurger control room design have rated high praise, but the studio room itself is also outstanding. Built within the existing frame of a huge stone barn and farmhouse complex, it has twenty-seven foot cathedral ceilings, large windows looking out on the surrounding woods (Harriman State Park is just up the hill), and two side walls built entirely of the original (reflective) stone. They've left it with a beautiful, natural feeling; rare in a studio so near to New York City and so professionally equipped. The natural light pouring into the studio, and the easy access to the outdoors, contributes to the open and comfortable feeling of the place. Still, Jay and Rich have produced a studio with a working philosophy. Attitudes of logical simplicity in the studio design, and openness to influence and change are its cornerstones.

When *MR&M* went there to visit, Jay was enjoying a day off and Rich was busy with finishing touches on the studio. They interrupted their afternoon for a conversation with *MR&M* about the new turns their work has taken, a little of their history, and how the studio functioned during the recent mix sessions. Michael Barry (now their chief engineer after working with them at Secret Sound in New York), also spoke with *MR&M* about his own views of Spyro Gyra, the new studio, and mix-engineering in general.



Modern Recording & Music: Your first well-known recording was independently released. What made you decide to do that?

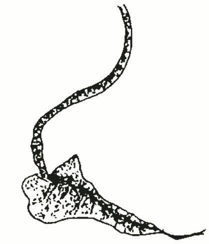
Jay Beckenstein: I had a band in the bars called Spyro Gyra, and I was getting together my own material along with Jeremy Wall. I met Rich and he had some previous experience in recording studios because he had done some things for CBS. He had been involved in some small way with Teo Macero. He was up in Buffalo and he found this really good deal on a recording studio and had this idea of a company that was basically a production company. I gravitated towards Rich because I had a good project, and ended up becoming his partner on the deal. We pooled our money and did five or six different projects; a country project, an R&B project, and *my* band, which was Spyro Gyra. For one reason or another, not all good reasons, none of that material was picked up by record companies when we went around with it. As we were nearing the end of our money and the end of our studio deal, we said to ourselves, 'At least we have a full album worth of Spyro Gyra; let's put it out and maybe we'll recoup a little bit of our money off the local fans.' That first Spryro Gyra album subsequently sold close to 200,000 copies.

MR&M: How did you do it? Did you get funding to do the promotion and distribution and the rest?

JB: No, it was our money; we matched money. What was amazing about that first thing was that we got to do everything you have to do in the record industry in microcosm. We started from writing the songs, to recording the songs and producing it and going through the whole mastering and quality control of making a record and then packaging a record and then going to radio stations and to distributors and to retail and advertising and, glory hallelujah at the end, the profits.

Rich Calandra: And it was the thing that we probably like doing the most, because it was just instrumental music, it was just blowing and playing new material. We *never* thought it would get off the ground.

JB: It was the project that we considered the least commercial.



RC: Yeah, it was twice as gratifying.

JB: Well, it wasn't even done for... it was done because I had this thing and we had spare time between doing the other serious projects, serious in that were supposed to support our company. And the irony is that the one that was just done for fun turned out to be the one that made us money. Fortunately, I get to do the music that's fun.

MR&M: When you were working in the studio back in your sixteen track days, were you developing ideas about how to record and how to mix that stick with you now?

RC: Sure, it was much more difficult.

JB: We developed our techniques early on, and at least as far as how we approach basics and overdubbing, it's remained fairly consistent. As time has gone on we've tried other things, and some of them we've accepted and some of them we've rejected. And we've gone some different ways on records, 'cause we've had the time and the money. But we did develop our basic technique during that first record.

RC: It wasn't only our first record that was recorded that way. Our biggest record has been *Morning Dance*, and that record was partially recorded in Rochester in another small studio; more than half, I would say, was done there. Everybody talks about the first one being the barn job, but *Morning Dance*, which was a Gold Record, and one of our best records, was also done that way. So, it can be done, you know.

MR&M: A lot of people are working on smaller budgets now, out of necessity. How do you set up sessions *now?* Do you write in the studio?

JB: No, I bring my arrangements into the studio completely charted out. I also bring in a demo tape I put together on four or eight track. I play keyboards, saxophone and I have a

YOU CAN AFFORD UP TO 900 MS OF STUDIO QUALITY DIGITAL DELAY!

Until just recently, state of the art digital sound processing was reserved exclusively for members of the studio elite or those willing to shell out some pretty hefty bucks. With the introduction of the Ibanez DM1000, you can now join the ranks of accomplished studio musicians and sound engineers. The DM1000 offers all the most desirable features of comparative units on the market today costing 2 to 3 times as much; including slapback, doubling, hard reverberation, flanging, chorusing effects, an infinite hold setting, and more at a price that lists for under $450.00.

Low noise and distortion, a multi-purpose input/output configuration (front and rear panels), plus a full studio quality bandwidth (at all settings) make the DM1000 ideal for live performances as well as studio and P.A. use.

Now you can afford studio quality digital delay. Stop by your Ibanez dealer and check out a DM1000.

Ibanez Tomorrow's Innovations

For a full color catalog send $2.00 to: IBANEZ, Dept. MR; P.O. Box 886, Bensalem, PA 19020
327 Broadway, Idaho Falls, ID 83401 17421 B. East Gale Ave., City of Industry, CA 91748
In Canada: 6355 Park Ave., Montreal, P.Q. H2V4H5

CIRCLE 25 ON READER SERVICE CARD

drum machine, so I demo the song, and have written parts for the musicians when I get into the studio.

MR&M: Are you going to have more time to experiment because of the new studio here?

JB: I sure hope so; that's one of the ideas behind having it here. I'd like to start experimenting with some non-Spyro Gyra approaches; maybe a more live approach, really try to do it all in one take, and then take it to the other extreme, a completely electronic do-it-myself approach. I'd like to experiment with those things. It's not that I know what direction I want to go in now, I have a lot of different directions I'd like to start dabbling in and the ones that are the most successful are the ones that I'm going to follow up.

MR&M: On *Incognito*, there are a lot of different feels from song to song. Of course, there are different writers.

JB: Yeah, well that's always helped. But I think on my four tunes I have four pretty different feels, too.

MR&M: Is that something you've always tried to do?

JB: I'm always looking to do something that I haven't done before. On *Incognito* for instance, 'Stripes' is really a direction I've never gone before, and 'Harbor Nights' is a little bit of a direction I've never gone before.

MR&M: Fans of yours have said, 'I hear Coltrane, I hear Sonny Rollins, I hear big bands, I hear Latin stuff...'

JB: I hope all of that is true and then some, because 'Old San Juan' is a big band Latin thing and 'Harbor Nights' is intimate jazz, done totally live. 'Stripes' is an Okey tune, you know it's got a real Texarkana kind of feel to it. That's coming from another direction entirely. And then, 'Soho Mojo' is straight R&B Motown.

MR&M: In most of the stuff that you do there's a lot of air, space, especially in the rhythm sections.

RC: Basically we put them together as raw as we can, with four-, five-piece rhythm sections at first. Sometimes we build on that. Sometimes there's no need to build on it, the tune starts to come through; sometimes we take it to the hilt, sometimes we take it to the hilt and we back off. That's basically the process we actually started with. Most of the time now, Jay's coming in really complete. But on a lot of the tunes, there's a melody and there's a rhythm. Sometimes some of the players, being so good, start writing the horn parts. We give a lot of freedom to the players.

JB: Of course, the players always put a lot into the music.

RC: They're creating. If you get too tight on it, it'll start sounding like a jingle session.

MR&M: When you're starting a session, how do you structure it?

JB: Actually, what we do is go for a live version, with the players aware that there are going to be overdubs, so they shouldn't overplay too much. I pilot all the solos and we try to give it the feel of a live session. If that session happens, that can be all we'll do. But usually we then take off whatever I did and just go back to the rhythm tracks. Start from there. Sometimes we leave the lead stuff on in those basic sessions. It's never just guys playing rhythm, I'm piloting everything. It's a live performance; we're trying to make things happen, we're not just laying down foundations. Often we just use the foundations from there and rebuild.

MR&M: How do you deal with songs that other people have written? Do you do the arrangements also?

JB: For the most part, the other

Korg's New SDD-3000 Programmable Digital Delay

Korg, world leader in electronic musical products, brings you a Programmable Digital Delay that really racks up points with recording engineers, pro sound men and musicians. It combines superb audio specs, long delay time and programming versatility at a price (under $1,500) that makes the SDD-3000 the most cost-effective signal processor on the market today.

IT'S GOT THE EFFECTS...

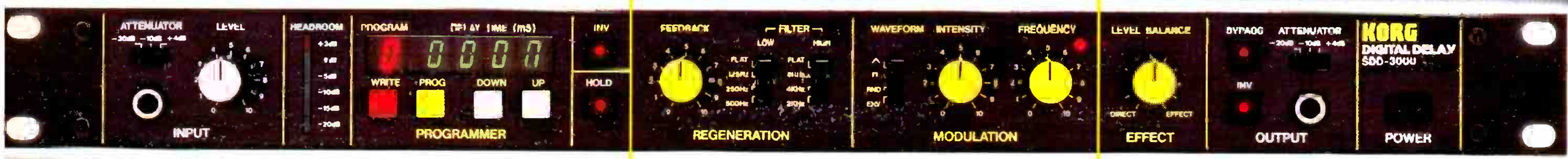
- Chorus
- Automatic Double Tracking
- Positive and Negative Flanging
- Resonant Flanging
- Vibrato
- Doppler Pitch Shift
- Short and Long Echoes
- Hard Reverb
- P. A. Phase Equalization
- Stereo Simulation
- Slapback
- Tunnel Effects
- Infinite Repeat

THE SPECS...

- **0-1023ms Delay**–with 20Hz to 17kHz frequency response at all delay settings
- **94dB Effect Dynamic Range** and,
- **110dB Direct Dynamic Range without** compander "breathing" effects
- **13 Bit Audio Conversion** for low distortion (0.03% typical)

AND THE FLEXIBILITY.

- **One Touch Programming**–9 different user changeable programs, plus manual setting. Programs can be edited temporarily or permanently and/or copied. Instant selection of programs via front panel or footswitches.
- **Expanded Modulation Section**–including 3 waveforms, envelope follower and separate external input.
- **A "Random" Waveform**–provides authentic doubling and other effects.
- **4-Position Low and High Cut Feedback Filtering**–for natural sounding echoes and unique special effects.
- **Versatile Stereo Capability**–includes both Direct and Sum & Difference outputs.
- **3-Position Input and Output Attenuation**–matches up to all types of instruments and sound systems.



KORG ...we put it all together

Exclusively distributed by Unicord
Westbury, N.Y. 11590

For SDD-3000 Digital Delay literature, write: Unicord, 89 Frost St., Westbury, N.Y. 11590. For demo record and full Korg catalog, enclose $3.00.

If you want a sophisticated delay at an unsophisticated price, see and hear the new SDD-3000 at your Korg dealer today. For name of nearest dealer, call: (800) 645-3188. In N.Y. (516) 333-9100.

CIRCLE 26 ON READER SERVICE CARD

people who write for this band are Tom Schuman and Jeremy Wall. Both of them have very strong ideas about the arrangements and Rich and I just try to help. Sometimes they'll come to us for advice and sometimes they'll have things all together. I like that because I think it gives the album that much more variety. Though there's a production team, depending on who the consumer is, the hierarchy changes a bit. Ultimately, Rich and I have final say, but the more we can just let it happen and not have to say anything, the better.

RC: Jay will bring in a tune and have some pretty specific ideas, too. Sometimes we do it, sometimes we can't. He's got this idea and he's gotta carry it through. It all turns into music, not to be defined, I think, as fusion.

JB: I've been attracted to so many different musical forms. I've found a lot of really neat stuff in all of them, and I try to apply some of that.

MR&M: Do you work hard on the sound of the sax?

JB: I work *very* hard on the sound of the sax, it's been my number one thing. You have an idea of what it's supposed to sound like; first you try to get it straight anyway, and then you can hone it down in the studio. It's all one long process and yes, we spend a lot of time on the saxophone.

MR&M: In order to fit it into some of the arrangements, do you have to alter the sound?

JB: Right from the start, the arrangements are built for saxophone. But there are all sorts of things you can do to the sound of the saxophone to fit it in, in a specific way. For the most part I'm just shooting to make it sound like a saxophone.

MR&M: But you have experimented with putting different kinds of sounds together; doubling the synthesizer with the sax...

JB: Sure, among a lot of things.

RC: Jay has a sound of his own, just like David Sanborn has a definite sound of his own. Jay definitely has that; I think he's had it since the beginning. I can identify it. If he's having a bad day, it's probably because he doesn't feel good or because he just got off an airplane, and then the sound goes away and I don't even care to deal with it as a producer on the other end because I know he'll have the same thought tomorrow rusted. You know, he's never at a loss. I hate to have him pinned to one thing as a producer, and yet sometimes he knows it was played well and is willing to keep it.

MR&M: It must be a relief to have the prospect of a setup where you can go in the studio and have the computer remember so much for you.

RC: I don't know, we don't call on the computer that much. We pretty much put that thing where we want it. When we get into the mixing stages it's quite helpful, because we all have different ideas about what something should flow like.

MR&M: So you do use a lot of different versions when you're mixing?

RC: Yeah, we make arrangements before we even get into a mix that we're not going to use a part of something like that.

JB: Now we've only used this computer one time, and that was in the last mix.

RC: But we've used other computers.

JB: That's true, but this was the most automated mix that we've had to deal with. I would come in and I'd sit there and work on my tunes and I'd go, 'Okay, which parts am I going to use, which parts aren't I going to use?' At *that* point, I bring Rich in, and the two of us decide on just a mute performance. If we have four solos, we tell the computer to mute the three we don't want so that all the parts we want now ride complete. So we get a complete mute performance, and then, all over again, Rich will start fiddling with sounds based on my decisions on what's being used and what's not being used. Then, we slowly gravitate towards our first real computer mix as far as levels go. And when we're very, very close to being happy with the mix, we do a level performance; from there on, we work an individual track at a time, honing it down.

MR&M: It sounds like the computer makes the process longer.

JB: It does. But it's a lot more exacting.

RC: It gives you more choices and it saves the time to find out. Like there're times when we've sat in studios, and we haven't used computers, and we'll just look at each other and say, 'Do we go for this?' We have to do this, we have to do that, it'll sway which way we go. Now with the computer, things are easier to get at.

JB: With the computer, you tell it to memorize it and you go to sleep and come back the next morning. The computer's doing all that stuff you used to do.

MR&M: Then you think of other things you want to do and change it again?

JB: Absolutely, but at least you're working on improving it. Considering it was really the first time we used our system, we got a great mix out of it.

RC: The Solid State Logic board itself is an incredible unit.

MR&M: You had worked with one before?

RC: Worked with one before but not to the extent of doing a full mix. And it was really great. I spend a lot of time on sounds. I'll just keep working them and then I'll hate them and I'll change them and I'll get them back again and then somebody'll come in with a different idea and I'll work on that for a while. It gets pretty tedious. This board makes it twice as fast, as far as getting you into the ballpark. Then, you start experimenting. Because it gets you to that place faster, you have more time for experiments.

MR&M: Have you worked with your engineer, Michael Barry, for a long time?

JB: Yeah, since *Morning Dance*. Only this last year has Michael worked just for us. Previous to that, he was the engineer that we worked with at Secret Sound Studio in New York. He was one of their chief engineers. But we've always had a special relationship with him, and when we had our own studio in the picture, he came to work for us exclusively.

MR&M: Did he have a lot to do with working out sounds in the beginning?

RC: Michael, I don't know, I get pretty good ideas of what sounds I hear. Michael is incredible as far as proving to you that you can make it a little bit better, so that sometimes you'll give him the time. He's such a creative engineer, you have to go, 'Wait a minute, there's a guy falling asleep out there, let's not do this any more.'

MR&M: Is it *his* rhythm section miking?

RC: It varies. I concentrate on bass and drums basically. I know the drummer, I know how he plays, I know who he is, I know the bass player, and we'll go for something immediately. If we brought another drummer in, we'd mic him differently, 'cause he hits harder or he does things a little differently.

MR&M: So there isn't a basic setup that you like.

JB: No. Actually, Rich can spend an entire day on drums for basics, and usually we book one day of basics, which is just for the bass player and the drummer and maybe, the guitarist and the pianist to come in. You book the studio for ten in the morning and the guitarist is not supposed to show up till four in the afternoon, because it takes four hours to mic and four hours to hone in. He goes crazy over the drums.

MR&M: There's so much air in the overhead setup, air in the cymbals, stereo separation.

RC: It depends. Some of the tunes have a lot of room sound. You'll have your overheads—the configuration will be different if you want a closer sound. I'll listen to the tune, I'll listen to the rhythm section, I'll go through it and start making alterations and telling them lies about us breaking down in the studio, and say, 'Give me a second, I gotta fix this,' and Michael will go out there and change something. Renting time stops you from having those luxuries. Now that we have our own studio, the other guy can go to dinner and he can do this and he can do that and we're not pressured by the time clock.

JB: My drummer in the live band, Eli, I'd like to have a matching set of his drums permanently in the studio. We'll tune it over a year, let alone four hours. I'm sure I'm going to spend months finding the spot I'm going to stand in in that room, just speaking acoustically, the place I should aim the saxophone. I'll probably spend years finding that perfect place.

MR&M: You do that a lot anyway?

JB: Yeah, some of the best sounds we've gotten have come playing the horn directly into my sneaker, flat at the floor and miking the bounce. It really got down to what shoe I was wearing. That's the truth.

RC: And it got down to, 'Jay, don't move an inch.'

JB: I'd chalk where my sneaker was on the floor and I'd have an interstice in the wood pattern that I'd aim the horn at. Yeah we've gone to those extremes. We've played in the bathroom at Secret Sound. We played in the hallway.

MR&M: You have a great silo out there next to the barn.

JB: We used it for a chamber during mixing.

MR&M: What's the room going to be like?

JB: Right now, it's live.

RC: Well, it's under construction as you can see. The control room's pretty close to finished. We mixed *Incognito* in there when it was even less finished than it is now. The main recording room has several areas that still have to be completed.

MR&M: But it *is* likely to be a live sort of room?

JB: Well, we have the option of it being a live room, because it has a very high ceiling and it has hard surfaces. Now what we have to do is, as we finish construction on it, deaden it to the point we like, or leave portions of it live and portions of it dead. We'll find those balances.

RC: It's good to be able to have it that flexible. It has twenty-seven foot ceilings and that's great height. They're slanted ceilings, and you can soften them, play towards them. The studio's actually fifty-seven feet long.

JB: And we have two huge, huge isolation booths. I mean, one isolation booth, you can put a whole band up there.

RC: I'm looking forward to setting mics up for Jay outside and having him play there; it's that quiet here sometimes.

JB: We're about ⅞ths of the way home. It's that last eighth that's taken a while.

The conversation continued, along more technical lines, with Michael Barry, chief engineer at Bear Tracks.

MR&M: What equipment do you have in the new studio?

Michael Barry: We have the Solid State Logic Series 5000 Console, we have two 24-track Studers, LA-2 Limiters, old Fairchild 660 Limiters, Publison digital delay, Super Prime Time Digital, EMT Reverb, all sorts of other limiter/compressors, Pultec outboard EQ. We like to go for those old 'tube' sounds.

MR&M: I noticed that your monitor system is a two-way system. Why is that?

MB: The persons we employed to design it, George Augspurger, built the control room around that concept, and we're giving it a try. It seems to be working out fine so far.

MR&M: What's the concept? Is it simply that there are so many crossover problems with three-way systems?

MB: Well, yeah, you get into all sorts of different types of problems with different crossover points. But with the two-way design you may not be getting the full frequency range that you might need. That's why they developed all these three-way, four-, or however-many way designs; to give each frequency range its own driver. But then you get into phase problems and all sorts of other compromises. It's all just a compromise.

MR&M: Did you end up altering the control room in any way while you were working there?

MB: While we were working there, yeah, we did a few things, but nothing to the basic construction. There are panels in the back of the control room that will change the room a bit, three different kinds with varying reflective and absorptive qualities—another design by Augspurger.

MR&M: Are you planning to use the studio for more than just Spyro Gyra projects?

MB: I hope so. We certainly have plenty of Spyro Gyra work, but I'd like to see some great people come through there because I think it's a great facility.

MR&M: The speakers in the monitor cabinets are made by whom?

MB: JBL on the low end and TAD on the high end.

MR&M: And what's the crossover point?

MB: About 800 hertz.

MR&M: Have you experimented with moving that around?

MB: It's a fixed crossover network. George Augspurger designed the cabinets, those are the speakers that he recommends with the cabinets, so...you've got to start someplace, otherwise you get confused.

MR&M: So if you wanted to change the sound in the room, you would first change the surfaces?

MB: Yes, to start with, sure. We've altered a little bit up in the front where the speakers are. That panel in the center, we're going to eventually have a [visual] monitor up there, and that alters the sound a bit. We've experimented with putting wood up there, and all sorts of things.

MR&M: You had a low-end hump from that?

MB: It actually helped that; it got flatter.

MR&M: How about running down the list of microphones?

MB: Neumanns, some Neumann tube mics, Telefunkens, AKGs (a wide range of those), all the other Sennheisers, the D12.

MR&M: Do you normally use old condensers?

MB: I prefer that. Not on everything, but on the saxophone that's pretty much our standard sound.

MR&M: Which mic?

MB: Oh, a lot of different things; AKG 451, or sometimes even a dynamic works, or a PZM® once in a while. We have the older ones, they're more like the original. I think they're four inch, four or five inch, but you know, you put them on a piece of wood or a piece of glass or something.

MR&M: Jay described a situation where he was aiming the sax at the floor, and he found that the best sound that he could get was when he was aiming right at his sneaker, and straight at a reflective floor.

MB: Yeah, for the soprano that seemed to work out. There are so many considerations, and that was just for that particular sound we wanted at the time. But yes, that's what we did. I won't deny it.

MR&M: Have you experimented a lot with that kind of thing on the sax especially?

MB: Yeah, we spend a lot of time getting the sax sounds; we spend a lot of time finding the right reed to use also. That's where it really originates.

MR&M: The studio itself looks like you're starting with a real reflective, very natural kind of space.

MB: Yeah, our philosophy is that you can always make it dead, but you can't have a dead room and make it live very easily. We all wanted to keep the stone, which I thought was a wonderful idea, and going with wood on the ceiling was something we all kind of decided on. Rather than just starting with a dead surface, we can always make panels and attach them to the walls. It seems like the industry goes through different phases in sound; it has sort of developed back into this live thing and people wanted to have a live room to record in. We've got that.

MR&M: You don't think the windows are going to give you any big reflection problems?

MB: Not at all. They're triple thermopane. And I think it's great having natural light in the studio! the musicians can look out the window. If they don't want to look at you, they can look out the window.

MR&M: Do you know how you're going to use that open loft area above the control room?

MB: No, we're leaving that open. We put steel beams to support the floor above the control room that are not attached to the control room, so it's an independent surface. We should be able to make plenty of noise up there, or vice versa; have open mics without picking up the control room.

MR&M: So the control room is totally floated.

MB: Yes, everything's going to be totally floated. The isobooth will be separate; we've been thinking about separating the floor in spots in the studio itself, half a hairline—just a sixteenth of an inch or something, like a kind of channel that would separate sections of the floor. It won't be anything noticeable that would obstruct movement across it; just enough so that they're not actually connected. Vibrations won't actually travel. That area above the control room should turn out to be really nice. It has a nice sort of ambience now. The room is certainly live! It's different for us, too, we're so used to places like Secret Sound.

MR&M: Why don't you describe your drum miking?

MB: It's pretty standard I think. D12 on the bass drum generally, Sennheiser 421s on the toms, overheads varying, 451 AKGs, (I like bright mics on the cymbals) and bright mics on the snare and high hat. We usually double mic the snare drum. Nothing unusual at all about them. I think that if the drums sound good in the studio, you really shouldn't have too many problems. If you use good mics, point them in the right direction...

MR&M: Well, how do you like to angle overheads, for instance?

MB: I like to angle them tight, at the cymbals.

MR&M: Do you use 451s frequently?

MB: I'd say so. They've got a kind of tight pattern, and they're bright mics; you get a little bit more control that way later on. You know, this is the Spyro sound too. I'm not saying there's any specific way to do this at all.

MR&M: No, but it gives clues about how a specific sound is arrived at.

MB: Yeah, but I'm skeptical about that because then people kind of develop the notion that that's the way it should be done, and there's *no* way that it *should* be done. There's absolutely no way...The way the Police approach a drum sound is obviously entirely different than the way we do it. We go for a tight sound, tight drums, tight snare. They go for more of a live sound.

MR&M: Do you mean yours include less leakage?

MB: Yeah, less leakage, but leakage can also be used to your benefit; it can enhance the sound. When you start getting into getting rid of leakage, you have to keep in mind a lot of phasing problems with multi-miking. You could get too much snare drum in the toms, and if you don't get the proper balances, you might have problems later on.

MR&M: Jay said earlier that Rich watches to make sure that the frequency ranges don't get cluttered up. How do you do that when you're in the basics, say on 'Old San Juan.' Did you know when you started out that it was going to be a dense arrangement?

MB: I assumed that, because I know the guys real well. I kind of know what to expect in a way, but even so, there *were* things done especially for 'Old San Juan,' on the basics, but they were all done in a series of days when we just kind of went in and got the sound and did it. It wasn't a whole day spent on getting the proper sound for 'Old San Juan,' it was more the general drum sound for the record and then fine-tuning it for each specific song. Some bands would go in (especially self-contained bands which have more time and more patience as individual people), to get the exact sound they want right from the start. Fleetwood Mac would get that snare drum exactly the way they'd want it, rather than getting it generally within the realm that you know you can work with later. You know, when you're working with a studio musician, you have to consider his time, his patience, all those other things. If you're self-contained, that's the only thing you're doing that day, so you don't have to worry.

MR&M: Did you find yourself wishing you could work on a particular sound?

MB: Oh, I think engineers *always* do that, always wish they could have the time to experiment; it's a challenge to work with studio musicians. You have to capture the magic quickly, while it's happening, which I think is fantastic; that's music, that's spontaneous stuff.

MR&M: How did you record the whistle on 'Old San Juan?'

MB: I've been experimenting with stereo miking. I do that a lot on percussion setups. We had a stereo microphone at Secret Sound, an AKG. I like to do (I think it's called) an MS kind of mic setup. We had the two capsules, one pointing up and one pointing down, or sideways or whatever, in omni. Or one in figure-eight and one in omni. That's actually what I did on Jay's whistle. I was kind of surprised because I had my doubts in the beginning, but it came out good.

MR&M: How do you actually record bass?

MB: Direct. I'd say 99 percent of the time, especially with these studio guys that know their sound.

MR&M: So you normally would just take it direct and not set up a mic in the room?

MB: Well, at Secret Sound we had room sizes to contend with, and leakage, so we couldn't have an open bass amp in the room. That presented a problem right there. But we've on occasion, during the mix or at some other point, taken the recorded signal and fed it back through an amplifier to the studio, or through some ambient speakers or something, and gotten the sound that way.

MR&M: How do you normally deal with the bass sound to make it fit into these dense mixes?

MB: Well, it's hard. We go for a tight bass drum sound. It seems that the bass drum fits a little under the bass on our records. Not under level-wise, but frequency wise. On this record, and *Freetime*, we got a better bass-to-bass drum perspective than we had earlier on.

MR&M: You like them to sound like distinct instruments and not to punch right in the same place?

MB: Yes, right, that's Spyro's general concept too, to keep things separate, but like you say, dense. Obviously, they do come together in the end, but you're still able to distinguish them and pick them out.

MR&M: Also, the wetness helps spatially. How did you place and separate things on 'Old San Juan'? It is almost a three-dimensional mix.

MB: Oh, we had a choice of four different reverbs, several different delays. We had the EMT, digital reverb, an AKG unit—we even tried that.

MR&M: A spring unit?

MB: Yeah, pretty surprising.

MR&M: What did you use that on?

MB: I'm not giving away everything!

MR&M: Oh come on!

MB: No, I'd be embarrassed. Nobody uses those any more, do they? We used an EMT 251, digital reverb, plus we used the room, and the silo.

MR&M: You used a lot of acoustical wetness for basics?

MB: It depends on the room. If you don't have the ambience in the room to work with, then you can't, and if it's a very noisy room where the leakage has a bad sound, a negative sound to what you want, then you can't use that either. On the other hand, the leakage might have a positive effect; you might be able to use it wonderfully, or the room might be live, where you can get the live sound you want. Or it might be big enough so you can separate the instruments physically from each other. And you can get the sound you want without having to worry about leakage at all. It just depends on the situation. Like I say, I did most of my work at Secret, had to deal with the situation there, so that's the only reason I say the things I do, out of the experience of that.

MR&M: How about piano?

MB: Acoustic piano; I like the sound on 'Old San Juan' a lot.

MR&M: What did you do?

MB: Two 414s.

MR&M: 90 degrees?

MB: Sort of near the hammers, 45 degrees.

MR&M: Aimed toward the low and high ends?

MB: Yeah, but near the hammers, because I knew that song was going to end up pretty dense. That was a fantastic pianist, Jorge Dalto. The way he hits and attacks the notes make them all come out so even, you don't have to fish around a lot.

MR&M: How's the computer system working out?

MB: It's wonderful for their method of mixing. It's very logical, straight ahead, and you can do a lot of recombining; just doing sections of a mix and then putting it with the other versions you liked. It's all pretty straight ahead and a lot easier than other types of computerized mixing.

MR&M: It seems to offer a lot more choices and thus take more time.

MB: Yes, that's true. The more time you have, the more decisions you have to make.

MR&M: What kind of choreography were you used to doing when you were doing all manual mixes?

MB: Well, I still mix manually—I do want to set that straight. I'd like to approach it that way. I hope that this automation fever doesn't really take over the industry because I think manual mix is wonderful...to maintain that human side of it.

MR&M: What things do you let the computer do? How do you divide the labor?

MB: I feel that you should set the mix up totally manually, get right up to the point where you want to do a take, then turn the computer on, make believe it's not there and do a take. Do your manual moves or whatever; you have that mix stored, so you can play it back if you like, or just move on. Try a few other things. You can approach it from that aspect or you can start right off with the automation and just mix the rhythm section with it. Then you can play that back and you can add in whatever other instruments you want, then play *that* back and add the next thing and the next thing, and then go back and fix whatever you didn't like.

MR&M: It can make the sequence of things you do in a mix slightly different?

MB: Well, yes and no. Even in a manual mix, you tend to go through each instrument separately and make sure it's okay, then fit it in with however you're building your mix. I just prefer the more manual approach to it. Plus, you know, I grew up on that system. I like to move all the faders all the time and feel out the mix as it's going. With the computer, every time you do that, it updates your new move. Then, the next time you play it and move the fader, it updates it again. It's just an ongoing thing, so you have to be more careful not to let the mix get away from you with the computer.

MR&M: You mean you have to be careful not to be too quick to make a move?

MB: Yeah, you have to be a little bit more sure about what you want to do. You can get back to what you had with the computer, but it just makes you think twice about doing it, rather than just doing it. Of all the computer systems, the SSL is wonderful. That's the concept they went on, that this shouldn't be a computer-oriented business, it should be a human-oriented business and the computer is there to aid you, not to be the center of attention and take things away from you. That's the concept they went with, and I think they've succeeded.

PROFILE

Philip Glass

by allan kozinn

Unusual as it may seem today, the idea of the classical composer as a pop star is hardly a new concept: In the 19th century, composers toured as virtuoso performers, unveiling their latest works in the concert hall much the same way today's pop performers put forth their latest music on LPs. And as with today's pop stars, these Romantic-era composers were bona fide celebrities, with masses of followers and, naturally, camps of detractors who preferred the musical and/or philosophical directions taken by rival composers. In our time, the classical music scene is not quite so volatile, and, in most cases, the contemporary composer has become a background figure who waits in the wings while performers (who, by and large, are more concerned with music of the past than of the present) take the spotlight.

While that remains the prevailing mode in the classical music mainstream, a group of composers who began pioneering a new compositional style in the mid-1960s—and who formed their own ensembles to play that music—are now not only edging their way into that mainstream, but are drawing their audiences from both the classical- and pop-oriented sides of the fence.

Among the most successful of these composer/performers is Philip Glass. A well-known figure in New York's avant-garde circles from the time he founded the Philip Glass Ensemble in 1968, Glass's ascent to his current prominence began in 1976, when he presented his first opera—a five-hour collaboration with director Robert Wilson, *Einstein on the Beach*—at the Metropolitan Opera. A Carnegie Hall concert soon followed, although Glass continued doing most of his performing in SoHo lofts, supporting himself meanwhile as a New York cab driver. In 1980, he scored an even more important success with his second opera, *Satyagraha*. First performed in Holland, where it was commissioned, *Satyagraha's* subject is the early years of Gandhi's non-violent resistance movement; but its text is drawn from the Bhagavad-Gita, an ancient Indian spiritual work, and it is sung not in English or Dutch, but in Sanskrit. The opera was received enthusiastically when it was presented at Artpark (near Buffalo) and at the Brooklyn Academy of Music, last season.

For the uninitiated, Glass's music is built of simple melodic materials—motifs derived from scales and arpeggios, and short cells built of small interval leaps—repeated continuously over a gradually shifting rhythmic and harmonic base. It is music that has been called "hypnotic," "trance," "pulse," "repetitive," or most commonly "minimalist,"—all labels that describe, however inadequately, a renegade musical style in which both the lavishly expressive impulses of Romanticism and the more stark, cerebral and dissonant stand of the mid-20th century serial composers are left behind in favor of continuously flowing textures. In the classical music world the style is a controversial one, which some critics view as too simplistic a response to 20th century compositional problems; in the pop world, though, Glass's music has found resonance in the music of Tangerine Dream, Davie Bowie, Brian Eno and Mike Oldfield, among others.

Glass's musical beginnings were traditional enough: Born in Baltimore in 1937, he began studying the flute at the Peabody Conservatory at age eight. At 19, he took a degree in mathematics and philosophy at the University of Chicago, and in 1962, he took his Masters in composition at Juillliard, where he wrote and even published several traditionally structured works. A composer-in-residence grant took him to Pittsburgh for a couple of years, and then, in 1964, he went to Paris on a Fulbright grant for further compositional studies with Nadia Boulanger (legendary as the teacher of Aaron Copland and the entire generation of American composers who came to prominence in the 1940s).

It was in Paris that the seeds of his new direction were sown. In 1965 he was hired by the Indian sitar virtuoso Ravi Shankar as an assistant during a filmscoring project, an experience that added new fuel to an interest in non-Western music that Glass had developed during a trip through North Africa sometime earlier. In 1966 he traveled to India and studied percussion techniques with tabla player Alla Rakha. Upon his return to New York in 1967 he began composing music in which the harmonic and rhythmic elements were stripped of the complexity developed in Western classical music over the last two centuries. He formed the nucleus of his ensemble, with two reed players embellishing the raspy sound he made on an electric Farfisa organ. Eventually the ensemble grew to include more reeds and another keyboard player as well as a sound mixer who was considered an integral part of the ensemble.

These two—keyboardist Michael Riesman and engineer Kurt Munkacsi—have become, next to Glass himself, the central figures in the group. Riesman, whose keyboard technique is stronger and more polished than Glass's, takes the more difficult keyboard parts and serves as a kind of conductor, keeping the ensemble on track during performances and playing a key role in the preparation of the group's recordings. Munkacsi, once a bassist for the rock band Ten Wheel Drive, has a similar but more technological function. On stage he sits at a mixing board at the center of the group, seeing to the mixes that both the audience and performers hear. Although he and Glass are technically co-producers of the Glass LPs in the studio, it is Munkacsi who serves as director during the recording and mixing. Glass's part in all this is, first and foremost, writing the music. He also plays keyboards—something he considers secondary—and he has the last word about how the recordings are mixed.

Glass's recording history has been strange. Unable to interest record companies in his music at first, he formed his own label, Chatham Square, which released several discs of his early works. In the early 1970s the British rock label Virgin Records signed him on and released two sections of the epic *Music in Twelve Parts* and *North Star*. A few years later a new

American label devoted to adventurous repertoire, Tomato Records, recorded Glass's *Einstein on the Beach* and two sections from *Dance*; but that label folded. In 1981, Glass was signed by CBS—a label that had an important composers' series in the 1950s and 1960s, but which hadn't been particularly involved with contemporary music since then. Early in 1982 CBS released *Glassworks*, a collection of short and attractive, although not particularly characteristic, pieces. The label plans to reissue *Einstein* and some of Glass's earlier recordings and will record *Satyagraha* when it is performed in Europe next year (as part of a trilogy with *Einstein* and his new opera, *Akhenaton)*.

In October, Glass and an expanded ensemble recorded their latest LP, which contains music from the play The Photographer. The sessions took place at the Greene Street Studio, located in the basement of a SoHo art gallery, The Exhibition Space. Glass has recorded several of his works there including *Einstein*, *North Star*, *Dance* and *Glassworks* and Munkacsi had a hand in designing it. *MR&M* dropped into the studio during the mixing session for the second of *The Photographer's* two acts.

Modern Recording & Music: You've had a strong following in new music circles almost since the start, and of course, other composers have worked along similar lines from about the same time you began your stylistic experiments. Why do you think it's been only recently that your music is attracting so much attention in the broader music world?

Philip Glass: I think maybe it's just that we did 12 years of good work before people began to notice. Apart from that, there's been an interesting feed-in and feedback between popular music and the kind of music I'm involved in. When I toured Europe in 1970 there were a lot of people in the audiences who were just kids then and who were starting their own bands. Some of them began working along similar ideas—that's the feed-in; the feedback has been from people like Bowie, Kraftwerk, and many others who have used the style, and thereby helped make my idiom current and accessible to an entirely different kind of audience. So my work appears to many people as a kind of classical counterpart to, say, the Who. But I've always thought of it as concert music. I've never tried to popularize it, and I think my audiences have always been able to see the seriousness with which I take the music. Perhaps that's slowed it [mass attention] down a bit, but I think that now, after all those years of exposure, I have an audience that is ready.

MR&M: It seems that Europe has been particularly important in your career—your early tours there did, as you say, have this far-reaching influence, and several of your major works were commissioned by European organizations and were first performed there.

PG: It's a funny kind of reverse snobbery the Europeans have: they like to discover Americans before the Americans do. You can see it in the seriousness with which they've taken jazz musicians, including Cecil Taylor, Ornette Coleman and Anthony Braxton, and it's something they take a certain pride in. It's been very advantageous for me—they discovered me in 1971, and I've played at European festivals from then on. So while on one hand it's a kind of put-down of the comparatively slow reaction of the American public to its own musical styles, it is something that is a source of great support to contemporary musicians, whether jazz or experimental. And in my case, as you point out, it still is: *Einstein* was first done in Europe; *Satyagraha* was commissioned by the city of Rotterdam, and my next opera will be done at Stuttgart. My chamber opera, *The Panther*, was also first performed in Europe.

MR&M: And I believe *The Photographer* was another Dutch commission?

PG: That's right. It was written for the Holland Festival and was first performed at the palace for the Queen last May (1982), and then it was put on elsewhere in Holland.

MR&M: The scoring is a little different for you—there are trumpets, trombones, the solo violin that resonates throughout the work.

PG: Yes, but those elements are also in my filmscore for *Koyaanisqatsi*, and there is a bit of brass in *Glassworks*. But of course, in terms of what people have heard on disc of mine, this will be a new effect, I suppose. They sound nice, don't they?

MR&M: The movement I heard had a strangely gothic sound—I suppose because of the bass and the low brass in it.

PG: It does, doesn't it? What *The Photographer* is, actually, is a theater piece based on the life of Eadweard Muybridge, the photographer. Muybridge shot his wife's lover and killed him. There was a big trial and he was acquitted—it was a very dramatic event in San Francisco in the 1880s. Now, we weren't able to put the entire score on the record. In performance, the first act was done as a play in which these events were depicted. The second act was a concert, and that's the part you've just heard. It's almost like a violin concerto, in a way. In fact, it's actually the first major violin music I've written since *Einstein* which had some solo violin music in it, and I was aware of that when I did it. The violinist who plays it is Paul Zukofsky, who also played in *Einstein*.

MR&M: He has also recorded a lengthy solo violin work of yours called *Strung Out*.

PG: That's right. He recorded that on his own label, CP², which is his vehicle for recording a lot of modern violin music. That piece is actually quite an old one, from 1967.

MR&M: You have established it as a matter of policy that you will not publish scores of your music because you don't want other ensembles to play it. Why is that?

PG: Well, actually, I've formed a publishing company on the advice of some friends, which makes it easier to collect royalties and to license, and so that I can negotiate publishing arrangements directly without going through somebody else. But the only cases where this is relevant—where I have to produce a published score for people's use—is with the theater pieces, the operas. *Satyagraha*, for instance, has been done in the Netherlands, at Artpark, in Brooklyn, and is planned for Stuttgart; and I've been approached by another European opera company and several schools. My goal in writing the piece was that it would be an opera that other companies *could* do, beyond the original production. But with my other works that is not relevant, and in terms of the chamber pieces, I have refused to allow performances by other groups. At first, this was for financial reasons—because although I was sole publisher of the music, I was making a much better living as the sole *performer* of it. Also, my feeling was that if someone in New Zealand wanted to hear the music, they would have to bring our ensemble there; whereas, if they could get the sheet music, then there'd be no need for us to come. Let me say that this is changing now, but it was certainly true for the first ten years the ensemble was together.

MR&M: Still, there are a few isolated examples of performances of your works by other people—besides Paul Zukofsky, there are a few recordings of your *Modern Love Waltz* (one on Robert Moran's Waltz Project LP, on Nonesuch; the other by the DaCapo Chamber Players, on CRI), and there was Mike Oldfield's recording of *North Star*, in a disco version.

PG: Well, the last one was a licensing deal: As you know, once you make a recording of a work, you must allow anyone else who wants to record it to do it. In that case, it's not so much that I allowed the disco version as that I couldn't prevent it. The others also break the mold in the sense that they are pieces I've written specifically for friends. The *Modern Love Waltz*, actually, is the violin music from *Einstein*. Here again, since it was solo violin music, and since it was recorded, that meant that anyone who could transcribe it from the LP and learn to play it could record it. By contrast, most of the ensemble music is too complex to take off the record, and those I am not planning to provide published parts for. So yes, there are all these exceptions that keep cropping up, but all I'm prepared to do now is make the theater works available. I am aware of the fact that there are performers out there who would like to play the music, and I do get requests; but unless they are people I went to school with, or other composers I know from the neighborhood, I tend not to even consider doing it. It's not fair, but what are you going to do?

MR&M: Returning to *The Photographer*, then—what will be on the record from the first act?

PG: There is a song from the first act on the record, as well as a dance and an instrumental version of the song. That last section, however, won't fit on the LP; but we have recorded it, and perhaps CBS will put the song out as a 45, with the instrumental version on the back. The second piece—the slow dance—didn't get recorded. So the record is by no means the complete music from *The Photographer*; but that's what happens when you make records—you begin to think about things in a different way. *Einstein* was about five hours long in performance, and it's just short of three hours on disc.

MR&M: How did you compress it? Did you speed things up, or make cuts?

PG: We cut things—or actually, we didn't cut any music, but accelerated the rate in which patterns change. In the case of *Photographer* I did a bit of cutting and trimming, but most of the material is there. Remember that when you do a stage piece on record, you no longer have the staging, and so the music may be redundant at times. These kinds of shortenings don't bother me; I'm simply adapting the medium of theater to the medium of records.

MR&M: That's funny, because there are people who are outraged when, for instance, a pianist cuts a repeat in a Beethoven Sonata.

PG: I'm not a purist in that way, although I've had that problem in reverse with conductors who don't want me to cut sections I think should be cut. I suppose when I'm no longer around *not* to protect my music, other people will defend it for me—far beyond what I probably would have done myself. These are practical things. and my approach to music and to recording is a very practical one.

MR&M: Let's talk about the recording process a bit. I'm curious about the division of responsibilities between you. Michael Riesman. and Kurt Munkacsi.

PG: We don't really play role games in the ensemble, but different people do different things well and my philosophy is to let people do what they do best. Michael has the best sense of timing in the group, so he generally starts the pieces and counts them off; in the rehearsals, he leads. He's worked with me for eight years and he knows what I want. When Kurt began performing with us, in 1971 or '72, we decided that the stage was the best place for him to hear what was happening. It was also the only place he could be that would allow an interaction between him and us—if something sounded wrong on our monitors, we could say something to him. At this point most of that contact is very much the same as you see in a string quartet performance where the closeness of the sound has to do with the group's physical proximity. If you put a quartet in isolation booths and give them headphones, their performance of the Beethoven quartets or whatever will not be the same. So our music is really chamber music—this is something I've always felt instinctively—and in a sense, that's how Kurt relates to the group. I'm probably the first ensemble leader who has used a soundman in that way—not as an extra whose job is to pump up the sound, but as an integral part of the creation of the sound, which is obviously his appropriate role. Again, in both recording and performing I try to have people do what they do best; they don't write the music, I don't try to make the sound mixes.

At this point, Kurt Munkacsi entered the conversation.

MR&M: Kurt, how do you see your responsibilities in the ensemble?

Kurt Munkacsi: Well, to go back to the start, the reason I began sitting onstage during the performances is that I had the theory that the amplification was really one big instrument, the same way a violin is; and I wanted to be close to that instrument so that I could hear it properly. When a violinist plays a concert, he plays the instrument, and the acoustics of the hall are the acoustics of the hall—he isn't altering them. So our theory was that we'd go in, get everything right, do a good concert performance, and the acoustics of the hall are what they are. Primarily, I'm interested in providing a good listening environment for the musicians, because I believe that even if the acoustics of a space are not so good, if the band is playing well and together, that will get across to the audience.

MR&M: Do you improvise to any extent, as a player might?

KM: My improvisation goes as far as turning someone up a bit if they're playing well, or turning them down if they're not.

MR&M: On stage you occasionally use headphones, but most of the time you work by ear; why is that?

KM: When I use headphones, it's because everyone in the band has his own monitor speaker, with its own mix coming through, and every now and then I have had to check all those mixes separately.

MR&M: Philip, you mentioned that a string quartet put in isolation booths wouldn't sound as lively as it would if the players were all together. I was surprised to hear, during the mixing session, that you use multi-track techniques to record your music. Not only is that extremely unusual for a "classical" ensemble, but in the case of your music, with its intricate cues and slow changes, I've been wondering how it all fits together and manages to sound like a live performance?

PG: It's taken us 12 years to figure out how to do this, and Kurt will tell you more about the process. Basically, these techniques give us a great flexibility. We can isolate a single part and get it just the way we want it without listening to the other parts or having them played over and over. There is a certain mystique to a live performance, it's true; but that's appropriate to a live performance. We are trying to develop a mystique of recorded performance, which is quite different. To go back a bit to your question about what each of us does within the context of the ensemble and in the recording studio, think of it this way: We're a sophisticated team of people who make records together. I write the music, Michael conducts it and Kurt records it, with Michael serving as editor.

But we are really into studio technology. We understand it, we know how to use it, and we make the most advanced use of it we can. The idea of going in and just hanging a microphone over the group, as an orchestra is recorded, does not appeal to us. We believe. We believe that this is the way music can and should be recorded.

MR&M: Well Kurt, how is it done—how is the music kept together?

KM: The first thing we do with a Philip Glass piece is figure out how long it is and at what tempo it should be played. Then we lay down a click track for however many minutes we need, against which Michael puts down a keyboard guide track that includes a rough outline of everybody's part. Finally, there's a vocal guide track—a track on which we say when each of the figures in Philip's piece changes: 'figure 1, repeat; figure 2,' etc.

MR&M: Then you overlay each instrument or section of instruments on separate tracks?

KM: On two separate tracks, actually. While we record section by section, I usually record each section with a stereo microphone—my favorite being a Neumann SM69, followed by AKG condenser mics. That way, when we mix the record, we have everything in stereo pairs, which gives us the acoustics of the room and the phase information. The electric instruments—keyboards and synthesizers—go directly into the board. In this studio we have a Trident TSM console, 24-track, and we use them all. Generally, I avoid putting important instrumental tracks at the edges, top and bottom; I keep those for the click and cue tracks.

MR&M: You also used a computer system in the mixdown?

KM: Yes, that's the Allison Automation System. Basically, you do the mix and make all your moves—bringing up channels when you want to, turning channels off—and the system remembers what you've done. So when you're ready to mix the 24-track tape down to a two track stereo master, you play the tape and the Allison Automation System remembers and repeats what you did during the previous mixing play-through. You can also update it as it goes along, if you've changed your mind about something.

MR&M: When you made the stereo master, a few minutes ago, you made two copies—one onto an analog machine, the other onto a JVC digital recorder using a U-Matic video cassette. What is your feeling about digital recording?

KM: I'm interested in it. I don't think it sounds any better than the best analog recording available. As you heard, we did a pretty extensive A/B test during the mastering, and there really was no audible difference. Of course, our original master tape was analog, so this might not have been the best way to tell—except that there was no difference in the quality of the copies we made on both analog and digital. Our analog recorder, which is the finest available, is the Studer A-80, using ½-inch tape at 30 ips.

MR&M: You also seem to be very interested in digital echo chambers.

KM: Yes, we use the EMT 251 digital echo chamber, and the Lexicon 224X. Digital echo sounds much nicer than any of the mechanical echo chambers we've used until now, and they are writing programs for these things that duplicate the acoustics of concert halls.

MR&M: How did you get involved with the technical side of recording?

KM: I started out as a rock'n'roll bass player, but eventually I just got more interested in the machinery of the recording studio and I decided to become an engineer. At that time the Institute for Audio Research was just starting, so I went there and got my technical background. At the same time I lucked out and got a job as a junior engineer working for John Lennon. He had five engineers working for him, and I was the junior, so I'd end up coiling the cables and all that. But that's how I got involved with Philip's group. What happened was that John Lennon had a 16-track remote recording van, and someone who worked with him knew Philip and liked his music, and offered to let Philip use the truck to record his first album—*Music in Changing Parts*. Being the junior engineer, and this being weird music—and it being on a Sunday and for free—I was the one sent to the session. I got very interested because John Lennon's big theory was that if you had a good hook in a song, and if you repeated it enough times, it would stay in people's memories. I was interested in this concept, and here was Philip repeating those hooks for 20 minutes at a time. So eventually I left John's studio and began working with Philip and the ensemble. I also began getting involved with a lot of avant-garde artists and began doing sound installations for people in the art world. Eventually I ended up building a recording studio in a basement—which was actually the first studio down here. When it was redesigned, I worked on the physical layout of the equipment, but by then I was no longer installing the equipment myself.

MR&M: You and Philip have also collaborated on other production projects. How did those come about?

KM: The first was Polyrock. Nancy Jefferies, at RCA, had become very interested in Philip's music and decided that she wanted Philip to produce a rock band to see what would come out of it. Philip agreed, so long as I would co-produce with him and they could find the right band, which took about a year. So Polyrock was it. What Philip did was write various keyboard parts and vocal arrangements; I did all the technical stuff. We split the production tasks much the same way we do with the Philip Glass Ensemble. Later on, I co-produced the first Waitresses LP with Chris Butler here in this studio, and Philip and I have just co-produced an EP for the Raybeats. Our next project will be producing a version of Carl Orff's *Carmina Burana* for Ray Manzarek, the keyboard player of the [now defunct] Doors. We ran into him and he had this idea for the project, and some tapes, and it all sounded very interesting. So we're going to do it this winter in California.

MR&M: Philip, what exactly do you do during the recording sessions?

PG: During the actual laying of tracks I'm always there, because occasionally a part will be unclear or have an error and I have to decide which notes should be played, or how things should be phrased, if there's a question. Obviously, I'm the only one who can decide that. During the mixing I'm there some of the time, but not all the time. I think it's very important to have someone who is *not* at the mixing sessions the whole time, but who can listen to it before it goes to the stereo master. Because sometimes, if they've been mixing for five hours, they'll miss things that someone walking in fresh will hear.

MR&M: Perhaps this is a bit less true with your most recent music, in which you've been expanding your instrumental and vocal textures, but for a long time the sound of the Farfisa organ was a central part of your ensemble's sound. Why a Farfisa?

PG: Well, when I began working with multiple keyboards my first players were Arthur Murphy and Steve Reich, and we used to rehearse at Arthur's loft because he had a piano. But it was clear that we needed a portable keyboard of some kind, and to tell the truth, I opened a copy of Buy Lines and found an advertisement for a Farfisa. I went to somebody's basement with knotty pine walls in Queens, and underneath the stairwell was the Farfisa. I still have that first one. Later, I found three others—all in basements in Queens with knotty pine walls—and all under the staircases. I got them for $150, and, when I was in Italy, had the factory refurbish them. Other than that, Kurt has re-balanced them a bit so that I fit in better with other instruments; he's also made the sound a bit cleaner. When I started using them people used to ask me why I use the Farfisa, with its lousy sound; but I've always liked that nasty, penetrating sharp sound, and we've spent a lot of energy finding the right amplification and speakers to reproduce the Farfisa's sound. We started out using Altec speakers, then Bose, and now we use JBLs.

MR&M: Kurt, what is the rest of your stage equipment?

KM: Basically I use a Soundcraft IS mixer, a pair of Neve stereo limiters, a pair of UREI graphic equalizers—10 band—and I'm in the process of getting some more limiter/compressors.

MR&M: Returning to the music itself, Philip; I noticed that in *Satyagraha* there seemed to be sections of music that were intrinsically emotional—a lot of the music is quite sad, and a lot expresses emotion in a very romantic way—something that I haven't heard in your chamber music. Do the chamber pieces also have those extra-musical associations for you?

PG: They don't seem to. I think you've pointed out something that's evolving—that I've tended to make a separation between theater music and concert music. The concert music remains what you might call non-programmatic, or abstract, while with the theater music, whether it's opera or dance, I've chosen to take on the problem of theatrical expectation in a very direct way. I think the kind of impact you look for in a large theater piece is different from what you might look for in *Music in Twelve Parts*. It would never occur to me, for instance, to write a finale for a concert work; but *Satyagraha* has two finales, and in *Einstein* there's one.

MR&M: You seem, in fact, to have been going in two directions in the last few years—not only the larger sonorities and more expressive writing in *Satyagraha* or the *Koyaanisqatsi* filmscore with its sweeping choral sections, but also the kind of pretty miniatures on the *Glassworks* LP. The first piece on that disk, "Opening," struck me as just a short step from Chopin's kind of modulations.

PG: It is a pretty piece, yes. On the other hand, I knew when we recorded it that there would be people disappointed to hear the record begin with a solo piano rather than a big ensemble piece. I suppose I could have stuck the piano piece in later, and opened with the ensemble; but I thought, why not zig instead of zagging.

MR&M: Since you mentioned it, *Glassworks* did come as a shock to a lot of people who have followed your music over the years, and who were surprised at the kind of music on the LP.

PG: Personally, I would have wanted to begin by recording *Satyagraha*, but CBS felt it would be a better idea to establish some kind of sales record with an album of short pieces first. But there's another issue here: I like to give myself permission to do things that are not expected of me. I would like to establish that there is a certain range of unpredictability in my music, and once I've established that, I can do whatever I want. It's hard to keep an overview of someone's work. I mean, what's that LP going to sound like in the context of 30 years' work? It's just going to be another record. I don't like to be caught in my own definitions, or to be trapped in descriptions of my music. You see, the language of the music I am writing is still evolving, and as far as I'm concerned, it allows for a wide range of expression. But you can't do it all at once—the style evolves one step at a time. I can say, however, that if I told you I was working on a new piece, you would have no idea what it would sound like. Yet, I could also say that you would absolutely recognize it as mine when you heard it. That's something, isn't it?

MR&M: They call it style, I think.

PG: But more than a personal style, it's a *range* of styles. There are other composers I'm often associated with whose pieces haven't changed in 10 or 15 years—which is okay, in a way; people often do that, and artists often do that in their painting. But it's not very impressive to *me*, and what *I* want is the freedom to write, say, a pretty solo piano piece, whether people like it or not. I probably could have gone on writing *Son of Einstein* all my life, and people would have been completely happy. It's much harder to write a piano piece when you know people are going to be saying, 'Why did he do that?'

MR&M: *Glassworks* is said to have been remixed specially for the cassette version—with Walkman players in mind. Is that true, or was it just record company hype?

✿ ✿ ✿

KM: It's true. More and more people are listening to Walkmen now, and I'm a big fan of personal stereos myself. What we did was we brought our Walkman headphones into the studio after the LP version was mixed, and remixed the tape so that it would be most effective when heard on headphones. We'll probably do the same with *The Photographer*. In a sense, it's a political statement: The record companies are making this fuss about how they're losing sales because people are taping the records. We don't believe that. We think they're losing sales because they're not putting out very interesting music; if they did record interesting music, and marketed it properly, it would sell. The Walkman mix was just a new approach to marketing the music, and it's gone very well. I should point out, however, that there was one problem. On the first batch, CBS reproduced them at too high a level, and they are distorted. So if your readers get one of those, they should return it to the store or to CBS and get a new one, which will sound better.

MR&M: Just a few questions on your new opera, *Akhenaton*—how's it coming?

PG: I've finished the first act, and it sounds terrific. We're doing something we've never done before, and that is that after I finish the scoring, we make a 24-track tape of it using synthesizers imitating the orchestral instruments. This will allow the directors and the designers to hear the piece and to know where, for instance, trumpet cues are. Normally, when they work on a new piece it's something they may have only heard in a piano reduction, which makes it harder to plan their end of the work. This will give them a rough idea of what the piece will sound like.

MR&M: Why *Akhenaton?* Why not the next installment of Gandhi?

PG: I've said jokingly that Gandhi could be an operatic trilogy; I dealt with the years up to 1917 in *Satyagraha*. A second opera could take the first years in India, up to the Salt March; and a third could go from there to his assassination. But for the time being, at least, I've completed the work I wanted to do with Gandhi, and in a way I feel that the period in South Africa portrayed in *Satyagraha* was the most creative period in his life—not that everything else, the working out of his ideals that came afterward, wasn't magnificent too, but his ideas and character evolved during those early years. By the time he arrived in India in 1918, he was already the historical person we know. With *Akhenaton*, I saw right away that he would work as an operatic figure. Like Gandhi, he was a rebel, and like Gandhi and Einstein, he was a complex character who changed the world he lived in and did it nonviolently. He did it by the force of his personality. In Akhenaton's case, of course, his rebellion had to do with monotheism, which clashed with the rest of Egypt's religious structure. I see the three characters as brothers, or kindred spirits.

MR&M: You tend to get involved with social causes in these works, don't you?

PG: That seems to be true, doesn't it? And I must say I find that surprising, because I don't think of myself as a politically motivated person. Yet, I must say that all these works come up with those issues. On the other hand, perhaps this is just my personal way of dealing with that. There was a time I tried to deny the political content of the Gandhi opera, but obviously I've had to come to terms with it. Certainly these are people who can command the stage, and who can occupy our attention and interest for an evening-long opera. And perhaps the kind of person capable of sustaining that interest is a person who makes a compelling social statement. Let's just say that I'm a reluctant activist whose tendencies in that direction are unleashed on the operatic stage.

CRAIG ANDERTON

Korg SDD-3000 Delay Line

Just when you thought this column had run out of new delay units to review, along comes the Korg SDD-3000. With a list price of $1495, it falls in the medium-to-higher price range already populated by several other rack mount delay lines; however, the SDD-3000 includes some unusual features, such as a programmer section which can memorize nine different sets of control settings, random and envelope follower control voltages, and easy matching to any type of signal levels. And the sound quality... well, read on and you'll see why this is a delay line well worth investigating.

WHAT IS IT? The SDD-3000 resembles most other delay lines of equivalent price, with the exception of the programmer section. The maximum available delay is 1,023 milliseconds (there is no memory expansion available for the unit). Bandwidth is down by 6 dB (voltage ratio) at 17 kHz, which gives a good, clean high end. It occupies a single rack space and has pleasing cosmetics. (Korg has certainly taken a giant step in that direction; it was only a few years ago that their equipment looked like something from an Army surplus store.)

As with previous reviews, we'll consider the SDD-3000 as a collection of different submodules, and look at each of these submodules in detail before evaluating the unit's overall performance.

- **Input stage**: Right off, the SDD-3000 gets extra points for having paralleled input jacks on the front and back. And, when you plug into the front jack, whatever is plugged into the rear jack is automatically disconnected. Thus you can have the input normalled to something like a tape track output, but you could also plug a guitar into the front jack for laying down a quick guitar part without re-patching. The only limitation is that the SDD-3000 is set up exclusively for unbalanced lines (although you can always treat half of a balanced line as an unbalanced line).

Once you're plugged in, you set the input range using a three-position switch. The three settings are labelled -30 dB (470 k input impedance, and intended for low-level instruments such as electromechanical pianos and guitars), -10 dB (47k input impedance, for semi-pro studios, synthesizers, and so on), and +4 dB (10k input impedance, for pro studio and PA equipment—note that changing the input impedance as well as the sensitivity for different ranges is a nice touch). This switch setting may be further fine-tuned with the input LEVEL control. A six-stage LED headroom indicator simplifies level setting, with five green LEDs indicating the "safe" range and a red LED indicating the onset of overload.

• **Delay time setting section:** This is physically part of the programmer section, but we will consider the delay time setting and programmer functions separately.

A four-digit, green, seven-segment LED display shows the delay time, which is selected with two pushbutton switches (one labelled DOWN, which decreases the delay time, and the other labelled UP, which increases the delay time). To go from one delay setting to the next, you hold down the appropriate button. Luckily, though, in order to keep you from growing old and gray while waiting to go from a short delay time to a long delay time, the SDD-3000 includes a "speed-up" circuit for fast forwarding (or reversing) through the delay range. When you first hold the button down, the delay time changes in 1 ms increments. As you continue to hold it down, the delay time changes in 10 ms increments, and then, 100 ms increments. It took me about 10 seconds to go from maximum to minimum delay, but if you want to hone in on a particular delay time, you'll have to fiddle with the buttons a bit (I'm pretty precise about selecting delay times, as I like them to be related to the rhythms of a given composition). Delay time selection was one aspect of the SDD-3000 which didn't knock me out; I still find the keypad entry of the now-defunct Echo/Digital Recorder to be the easiest way to specify delay time. I shouldn't complain too much, I suppose, but after playing with the PCM-42, I feel that there are better ways to put two buttons to use than what Korg decided to do with the SDD-3000.

Incidentally, there's also a HOLD illuminated pushbutton switch in the delay time setting section which lets you store and repeat whatever sounds are currently held in the delay line. I think we've covered the subject of Hold switch applications so thoroughly in the last few reviews that we'll just say that it provides "solid-state tape loop" effects.

• **Regeneration section:** This section includes a FEEDBACK (feedback level) control, four-position HIGH FILTER switch (flat, 8 kHz, 4 kHz, and 2 kHz turnover frequencies), four position LOW POSITION switch (flat, 125 Hz, 250 Hz, and 500 Hz turnover frequencies), and an INVERT illuminated pushbutton (this alters the polarity of the regenerated signal, and is most commonly used to alter the tonality of flanging and chorusing effects). Since echoes in natural environments lose higher frequencies at a faster rate than lower frequencies, the high filter helps you obtain more realistic-sounding echo effects. Also note that the filter is in the feedback path, so each successive echo has less and less high-frequency content. This insures that newer echoes are more prominent in the final output than older echoes, which do not just fade away, but "filter away" as well.

The low-filter switch is great for eliminating the boominess which can occur with high-resonance flanging or chorusing settings. You can also create special effects by making successive echoes BRIGHTER (the opposite of using the high filter); this is particularly effective for emphasizing sibilants on vocals to help cut through a track a little better.

• **Modulation section:** This section has an INTENSITY control to vary the amount of modulation, a four position WAVEFORM switch, and a FREQUENCY control (with an associated LED that flashes at the LFO rate). Available waveforms include the standard triangle wave, the less standard square wave, the even less standard envelope follower, and the extremely rare "random" waveform. That's the good news. The bad news is that, like many other digital delays, the modulation range is limited—in the case of the SDD-3000, 2:1. That's alright for chorusing and echo, but for flanging and some special effects I still think an analog delay (with its inherently wider sweep range) is the way to go.

• **Output section:** First, there's an EFFECT control which selects the blend of delayed ('effect') and straight ('direct') sounds. There's also a front panel output jack which carries the direct and in-phase effect sounds; a three position attenuator switch labelled -30, -10, and +4 dB (this allows you to alter the output level present at the front panel jack to properly match various devices—guitar amps, semi-pro recorders, consoles, etc.); a clickless BYPASS illuminated pushbutton which lets you bring the effect in and out, and an INVERT illuminated pushbutton (which is not clickless, but then again, neither is the one in the regeneration section) that changes the in-phase delayed signal appearing at the output to an out-of-phase delayed signal.

• **The jacks on the back:** As mentioned earlier, there is an input jack which parallels the one on the front. There's also an output jack (called +MIX/MONO) which carries the mix of direct and in-phase effect sounds selected via the EFFECT control (in-phase assumes that the output section INVERT pushbutton is in the non-inverting mode). Note that unlike the front panel output jack, this output is not affected by the front panel attenuator switch, and is fixed at a nominal +4 dB. There are two other output jacks: one is a direct out which carries the direct sound only, while the other (-MIX) carries a mix of the direct sound plus out-of-phase effect sounds. Both of these deliver a nominal +4 dB signal level. By feeding the -MIX output to one channel and the +MIX/MONO to the opposite channel of a stereo field, you will hear synthesized stereo effects. If this stereo signal is played back in mono, the effect drops out and you are left with the straight sound only. Thus, synthesized stereo is more useful on-stage than in the studio.

There are also some control inputs which I'm sure were put there for reasons other than scoring brownie points with me. There are two jacks, intended for footswitches, which let you step through the various stored programs (one step upward—program #0, program #1, program #2, etc. every time the hot connection goes to ground, while the other steps downward when the hot connection goes to ground). There is also a jack for a Hold footswitch (a valuable feature if you plan on using the Hold function), another jack for a remote bypass switch, and a delay modulation control voltage input which accepts a 0 to +5V control voltage. Note, however, that you can still only sweep a 2:1 range regardless of what kind of control voltage you send into this jack.

- **Programmer section:** This section includes two switches, WRITE and PROGRAM; also, to the left of the four LEDs which indicate delay time is a red seven-segment readout labelled PROGRAM. This readout indicates 0 through 9, with 0 meaning that the front panel controls define the sound, while 1-9 are programs stored in the unit's memory. In case you're not familiar with programmable effects, what this means is that once you've found a particularly pleasing setting on the unit, you can store those control settings in the unit's memory, and assign this "program" a number. So, if you come up with a great chorus sound, you can store it as, say, program #7. Then, whenever you punch up #7, you'll hear that same chorus sound. The parameters which may be programmed are the delay time, all regeneration section controls, all modulation section controls, and the effect balance control. Once you've called up a program, the front panel knobs allow you to further modify (edit) the sound just in case the program needs to be changed somewhat for a specific application. A decimal point in the program readout shows that a patch has been edited from the original settings. If you want to get the unedited setting back, simply step through the programs until you again reach the program you were editing. The original sound will return.
- **Other controls:** The only control we haven't mentioned is the power on-off switch...there, we've mentioned it.

PRE-FLIGHT FOR THE SDD-3000. If you're into synthesized stereo, feed the outputs as described above to a stereo setup. Otherwise, you can treat the unit like a standard monaural effects unit and use the front panel jacks. Plug your instrument into the input jack and run a cord from the appropriate output jack (either the one on the front panel if you need to use the output attenuator switch or the +MIX/MONO jack on the back if you have a +4 dB setup) to your amplifier or other monitoring system.

I tried the SDD-3000 with a variety of signal sources

and levels—from unpreamped guitar to extremely high level signals—and had no level-matching problems whatsoever. This is one piece of equipment that tries to cover all the bases. With the exception of balanced lines, they've succeeded admirably.

If you're into stereo spreading techniques using delay lines, the direct jack comes into play. This would be panned to one channel, while the +MIX/MONO would go to the opposite channel. Note that the blend of effect and direct should be mostly effect for this kind of patch. The stereo effects provided by this patching scheme do not disappear in mono, although at short delay times, the tonality of the signal may change.

Now is probably a good time to point out that the SDD-3000's manual is excellent. It not only tells how to work the controls, but gives a good theory of operation which is neither so technical as to be overwhelming nor so simplistic as to be useless. There are also a number of patch charts which give different control settings, but these are really just points of departure; you should definitely experiment. And since the manual is one of those four language jobs, you can brush up on your high school French as you work your way around the control settings.

USING THE PROGRAMMER. Storing a sound is easy. First, you use the PROGRAM button to step through the program numbers until you hit #0. Next, you set the controls for the desired sound. Suppose you want to store this program as sound #4. When you've got the sound you want, hold down the WRITE button and use the PROGRAM button to step through the program numbers until you reach #4. Then, while holding down WRITE, press the PROGRAM button once. This stores the control settings as program #4. I should also mention that you can step through the programs by holding down the program button and pressing the "Up" or "Down" button normally associated with delay time. So, if you're currently in program #8 and want to get to program #7, rather than stepping through 9, 0, 1...7 to reach the sound you want, you could simply hold down PROGRAM (which would advance you to #9) and then hit DOWN two times to go backwards through #8 until you hit #7. Frankly, I find it easier just to use the PROGRAM button most of the time, but there are times when this option is useful.

By the way, it's worth noting that the SDD-3000 includes battery-backup so that the settings are stored more or less indefinitely. However, the manual does recommend turning on the power for a few hours every three months to keep the battery charged. Of course, who would spend $1495 for a delay line and *not* use it at least once every three months?)

DELAY LINE EVALUATION. Ah yes, you might say, all these toys and games and programmers are certainly very nice and high-tech, *but what about the sound?* To me, this is what was most impressive about the SDD-3000. It is the quietest digital delay line I have tested; they spec it at -88 dB S/N ratio, and I believe it. The sonic accuracy is superb—quiet, good definition, no noticeable glitches or artifacts, and a clean high end. Turning up the regeneration and kicking in the low filter for the brightest sound produced some noise, but I had to really work just to get a tiny bit of hiss out of the thing.

The feedback section sounds just fine, and since the headroom indicator also takes the feedback signal into account, it's easy to see if you're getting too adventurous with the overall amount of feedback. I particularly like the inclusion of the high and low filters; there should be a law that says every delay which does not include "loop" jacks in the regeneration path should include some kind of filtering option.

Moving along to the modulation section, I liked everything about it except that I would have liked the LFO to go a little slower (it seems that very few manufacturers have discovered the usefulness of extremely low LFO frequencies with ambient and new age music), and I would have liked a slew control for the envelope follower. Actually, I'm grateful that the envelope follower is there at all, because it is useful. But it would be more useful if you could slow down the response a bit so that the envelope follower followed not just the dynamics of each note, but the *overall* dynamics of your playing. Also remember that a 2:1 sweep range is not really enough for really dramatic envelope follower effects.

The random waveform is a little strange, but I received a very early model and was told by one of their engineers that subsequent models include a mod to make the random sound more musically useful. (To see if you have one of the older units, open up the SDD-3000 and examine R167 on the top circuit board. If it is a 10k resistor, change it to 47k). I still found the unmodified random control voltage valid for many applications, however.

EVALUATING THE PROGRAMMER. I wonder about the cost-effectiveness of including a programmer section; 9 programs is not a lot, certainly not enough for a studio to program all the sounds it will need. For live use, a programmer makes more sense although, even then, I would think that the average band would require more than 9 settings to get through a standard set. I suppose the best use of this feature would be to store a flanger sound, chorus sound, echo sound, doubler sound, a couple of gimmick sounds, and then make any needed fine tunings to these basic settings with the front panel controls. If the programmer only adds a few bucks to the list price and Korg figured they might as well throw it in, that's okay by me... I'll never object to something programmable, even if it is somewhat limited. But if the programmer is a significant part of the price, they should have gone for the additional expense necessary to store a larger number of programs.

OVERALL EVALUATION. Despite the above comments, I should emphasize that programmability can indeed save time when used intelligently... my main questions are: How much does this add to the price, and, is the capability truly worth the extra cost? Also, I'm not really that pleased with the delay time selection process. Granted, that's not the biggest deal in the world, but it was occasionally frustrating.

What *is* a big deal is sound quality. Regular readers of this column know that I'm real picky about noise, and, as far as I'm concerned, the SDD-3000 is the closest I've heard yet to being truly noise-free. What's more, it has excellent high frequency response (and extremely low distortion) to go along with the low noise. Overall, the SDD-3000 is not only technically impressive, but because of the superb use of technology, is musically impressive as well. If you don't mind stretching your budget a bit, the SDD-3000 delivers stunning sound, convenience features, and easy operation.

Ambient Sound

By Len Feldman

Unmasking The Audio

By now, I've had an opportunity to check out three different CD (compact disc) digital record players in my lab and listening room. From my experiments, I have come to an important conclusion which concerns everyone involved in recording. In fact, what I'm going to talk about here will be of concern to anyone who is involved in tape recording, professional or otherwise. I've concluded that a lot of the sins that we all got away with (and still do) in the analog audio domain are not going to be forgiven in the world of digital audio.

Let me explain. One of the first digital audio discs I ever heard contained two different symphonies. (Compact digital discs are able to hold one hour of recorded material on a side.) The first selection sounded great—as good as I've come to expect from the new digital disc format. Then came the second selection. It was a recording of Schubert's unfinished symphony, the 8th. As soon as I heard the first few notes I knew that something was very wrong. I heard an unmistakable tape hiss as well as definite compression and even some pumping and breathing. How could this be? Digital discs aren't supposed to exhibit any of these undesirable qualities.

Well, if you haven't already guessed, the answer is simple. The Schubert work had not been *mastered* on a digital tape recorder; it was a common, garden variety analog master tape. Still, the label on which this recording had originally been issued was a highly respected one (I'm not going to embarrass them by telling you who it was). I would guess that the tape hiss level was a good 65 to 70 dB below the peak musical recorded levels of this recording. Yet during the quiet passages (and even during the not-so-quiet ones) the tape hiss came barrelling through, clearly audible not only to me but to anyone else who listened to this "digital" disc. The contrast between the first selection on the disc and the second was truly startling. It didn't take an expert to quickly determine which one had been digitally mastered and which was simply an analog tape master transferred to the new digital format.

I decided to carry the experiment a bit further. I went out and acquired the analog version of the same performance of the Schubert symphony and played it on my regular turntable, feeding the same amplifier and loudspeaker system. This recording sounded fine—or at least as fine as any analog disc recording can sound after you've been listening to true digital discs for a while; the familiar surface noise was there, audible from the moment the stylus touched down on the lead-in groove. But somehow the tape hiss which had bothered me so much when I listened to the compact digital disc was magically gone! Of course, as anyone who is the least bit familiar with the science of psychoacoustics knows, the tape hiss wasn't really gone at all. It was simply masked by that ever-present surface noise. I wasn't as conscious of the compression which I had complained about earlier, either. I'm not quite sure why, but I suspect that now that the expected background surface noise was "in place" where I expected it to be, the rest of the balance of the music seemed "right." In other words, we are all so accustomed to hearing music after it's been processed by the tricks of the recording engineering trade that we don't even notice that anything's wrong.

When you take away one of those extraneous elements (in this case, the underlying level of surface noise), we apparently sense all the other disparities between a recorded performance and the real, live experience more acutely. Not only does the tape hiss of an analog master tape recording become "unmasked," but in a figurative sense, all the other deficiencies of the recording business become unmasked as well.

Why They Complain About "Digital Sound"

All of which brings me to a touchy subject: the recent bad-mouthing that some self-appointed audio experts have been engaged in for some time now. If you follow audio literature at all, you've probably read more than once that this expert or that expert is convinced there's something basically wrong with digital sound. One expert tells you that it's overly bright. Another insists that he can hear problems with phase relationships. Still another hears all manner of anti-aliasing filter ringing, while yet a fourth expert will give you any number of reasons why 44 kHz (or 48 kHz, or 50 kHz) digital sampling just isn't enough to do the job.

Like many of you, I listened patiently to these experts for quite a while. After all, they had had quite a lot of experience with listening to digital sound, whereas I had not. Well, now I have, so I'm going to add my own measure of "expertise" to the debate. My feeling is that some recording engineers had better relearn their trade if they plan to survive in the world of digital audio. Oh, and once and for all, let's not confuse the world of digital audio with the world of digitally-mastered audio. In making master recordings for the latter type of disc (whether on an analog or a digital tape recorder), the things you learned in recording school (or in your apprenticeship) are perfectly valid. Fixing a recording in the mix is just fine. Equalizing on the console in post-production editing and mix-down is OK, too, as is rearranging the balance and levels of every one of those separately-recorded tracks to suit your taste (or the taste of the artist who has been conditioned to such techniques, or even the taste of the producer who thinks he or she knows what will sell).

What I'm specifically talking about is the reeducation process that you will have to undergo once you start preparing master tapes for ultimate compact digital disc production. Here cover-ups won't work—and neither will quick-fixes. The audio you play back digitally from a compact digital disc is completely unmasked; all the faking shows through. You can no longer hide behind a veil of surface noise or of simpler forms of distortion which tend to cover up the more subtle kinds of distortion.

Will you be able to make this transition? Certainly! I've now heard enough superbly-engineered digital discs to convince me that there are already several recording engineers out there who have figured out what it takes to come up with the kind of master digital tape recording that will sound great when it's transferred to mass-produced compact digital discs. I don't profess to know much about recording engineering (I come from the audio hardware rather than the software discipline), but I can tell you which of the couple of dozen CD discs sounded good to me. They were the ones which had the least amount of signal processing added to them; the absence of any post production equalization; the most basic microphone placement. In short, they were the ones which allowed the digital medium to exhibit its amazing transparency which, despite the criticisms of the diehards, is after all the greatest positive attribute of the coming digital disc format.

Could it be that the coming of digital compact discs is actually going to *simplify* the job of the recording engineer rather than complicate it? And, if that is so, is it just remotely possible that some of the severest critics of digital sound are really subconsciously afraid of what this new, honest kind of recorded sound will do to their job security? I know a young FM radio station producer in Chicago who, armed with nothing more than a Sony PCM-F1 Digital Audio Processor, a matching portable video tape deck, and a couple of high quality mics, does weekly remote recordings of symphony concerts which are subsequently aired over his station with little or no further editing required. No 16-, 32- or 48-track consoles required here. Just reasonable placement of mics in the classic stereo manner, proper adjustment of peak recording levels so they don't exceed the 16-bit word capability of the digital processor, and that's it. Kind of makes you wonder, doesn't it?

LEN FELDMAN

Phase Linear Pro 700 Power Amplifier

General Description: The Phase Linear Pro 700 is a high-powered dual channel amplifier intended for professional high-fidelity sound reinforcement work. While Phase Linear is perhaps best known for their home stereo and hi-fi products, don't get the idea that the Pro 700 is just a home hi-fi amp with quarter-inch phone jacks for inputs. Its rugged construction, built-in ventilation fans and thermal protection circuits go far beyond the usual "home" amplifier in terms of reliable long term operation.

Of rack-mount width, the Pro 700 weighs in at 60 pounds. The two identical power amplifier channels are equipped with fixed input level attenuators whose front-panel knobs are calibrated in dBm required at the input for rated power output (360 watts per channel into 8 ohm loads). For example, when the attenuator is set at 6 dBm, an output of 1.5 volts from your mixing board or any other souce (electronic crossover, etc.) will result in a 360 watt output to the speaker load. Two LED power meters, configured to look more like conventional mechanical meter movements than like electronic indicators, simultaneously display both output voltage and power levels by sequential lighting of a 36-segment LED array. The meters are peak responding and have a readable dynamic range in excess of 40 dB with power calibration set for 8 ohms. A meter range switch changes meter sensitivity by 20 dB. A power on/off switch completes the front panel layout.

The rear panel of the Pro 700 has unbalanced input phone jacks, color coded 5-way output terminals and supply and line fuses all clustered at one end of the amplifier. The rest of the panel area is given over to a pair of ventilating fans (which are mounted directly behind the twenty power output transistors) and a huge power transformer which powers both channels of the amplifier.

Circuit Highlights: The low level input stage of each amplifier employs a wideband low-noise FET input IC op amp which provides high open-loop gain. A transistor operating as a voltage level shifter couples the output of this op amp to a common-emitter amplifier, which is biased for class A operation and is capable of swinging the full power supply voltage of 200 volts, providing the final voltage gain of the amplifier. The output stage forms a unity voltage gain buffer, capable of delivering the required current.

The output stage of each channel employs ten high-current, high-voltage silicon power transistors arranged in a complementary format and biased for class AB operation. Voltage and current levels of the output transistors are monitored by the built-in protection circuit. This protection circuit consists of two transistors, one operating when the amplifier output is positive, the other when the amplifier output is negative. If voltage and current levels are excessive, the protection transistors will conduct to divert current drive entering the pre-drive transistors, thereby limiting the current flow in the output stage to a safe value.

Thermal protection is also incorporated, causing the amplifier to turn off when excessive temperatures are reached. After cooling off, the amplifier will automatically turn itself on again.

Test Results: In *MR&M's* lab tests, the Phase Linear Pro 700 met or exceeded just about all of its published specifications with the exception of signal-to-noise, where we measured 90 dB unweighted and 96 dB "A"-weighted. For its rated distortion of 0.09 percent, the amplifier delivered over 400 watts per channel of mid-frequency audio power into 8-ohm loads. At 4 ohms, under the same test conditions, the amplifier put out in excess of 600 watts per channel. Running that kind of power level for more than a few moments triggered the thermal protection circuits even though the built-in fans were doing their best to keep things cool; Phase Linear does not recommend driving this amplifier into impedances any lower than 4 ohms.

When we reduced the output level to the rated value of 360 watts per channel (8-ohm loads), harmonic distortion dropped further, down to 0.017 percent at mid-frequencies. At the frequency extremes of 20 Hz and 20 kHz, harmonic distortion was still a bit lower than the claimed 0.09 percent, measuring 0.08 percent. Other forms of distortion such as CCIF IM and IHF IM were all so low as to be audibly insignificant, though for some unexplained reason SMPTE-IM distortion read a bit higher than claimed. A summary of all test results will be found in our Table of VITAL STATISTICS.

General Info: Dimensions are 19 inches wide, 7 inches high, 12 inches deep. Weight is 60 pounds. Price: $1395.00.

Individual Comment: I have always had a great deal of respect for Phase Linear products, though my experience with the company had, until now, been largely confined to their work in the consumer audio field. Having operated and listened to the Pro 700, that admiration now extends to their professional amplifiers as well. Here is a no-nonsense amplifier that is rugged enough to withstand the rigors of heavy usage in high-powered sound reinforcement work, while at the same time delivering the kind of accurate sound that will please even the most critical hi-fi types. I was particularly pleased with the topology of the metering arrangement, which was both easy to read and easy to interpret. One feature that was particularly welcome was the flashing of the top three LEDs whenever rated output or clipping was approached; sort of a gentle warning sign for the operator of the system that he or she had better back off a bit to avoid clipping distortion.

Compared with similarly powered amplifiers that we have measured in recent months, the Phase Linear Pro 700 is something of a bargain, a matter which no professional in the audio field can afford to ignore.

PHASE LINEAR PRO 700: Vital Statistics

SPECIFICATION	MANUFACTURER'S CLAIM	MR MEASURED
Continuous power for rated THD, W. (8 ohms, 1 kHz)	360	408
Continuous power for rated THD, W. (4 ohms, 1 kHz)	550	612
FTC rated power (20 Hz to 20 kHz), W.	360	378
THD at rated output, 1 kHz (8 ohms) %	.09	.017
THD at rated output, 1 khz (4 ohms) %	.09	.06
THD at rated output, 20 Hz (8 ohms) %	.09	.08
THD at rated output, 20 kHz (8 ohms) %	.09	.08
IM distortion, rated output, SMPTE %	.09	.047
IM distortion, rated output, CCIF %	N/A	.009
IM distortion, rated output IHF %	N/A	.022
Frequency response, @1 W, Hz-kHz, for -1 dB	12-40	13-37
S/N Re: 1W, "A" weighted, IHF, dB	N/A	75
S/N Re: rated output, "A"-weighted dB	100	96
Dynamic headroom, IHF, dB	N/A	1.4
Damping factor, @ 50 Hz	330 @ 1 kHz	Confirmed
IHF input sensitivity, volts	N/A	80 mV
Input sensitivity re rated output, volts	1.5	1.5
Slew rate (volts/microsecond)	20	20
Power consumption (watts) idling	N/A	60
Power consumption, max. (watts)	1200	1300
Dimensions (w" x h" x d")	19 x 7 x 12	Confirmed
Net weight (pounds)	60	Confirmed

Suggested retail price: $1395.00

CIRCLE 50 ON READER SERVICE CARD

Otari MX-5050-MKIII-8 Professional Reel-to-Reel Recorder

General Description: The Otari MX-5050-MKIII-8 recorder is a professional quality, two-speed 8-track tape recorder/reproducer which accommodates ½-inch wide tape and operates at either 15 or 7½ ips. Among its many professional features are: selective reproduction (SEL/REP), automatic motion sensing control, an edit control that permits tape spilling, dynamic braking, an electronic real-time tape counter, an adjustable cueing control for audible monitoring in fast-forward and rewind, a dual-frequency test and cue-tone oscillator, adjustable bias, equalization and level controls, XLR connectors for line inputs and line outputs, standby mode for ease of multi-channel recording, remote controllable electronics, a VU meter with peak indicator for each channel, selecting switches for input and output levels, and a memory stop for automatically stopping the tape at a desired point during playback.

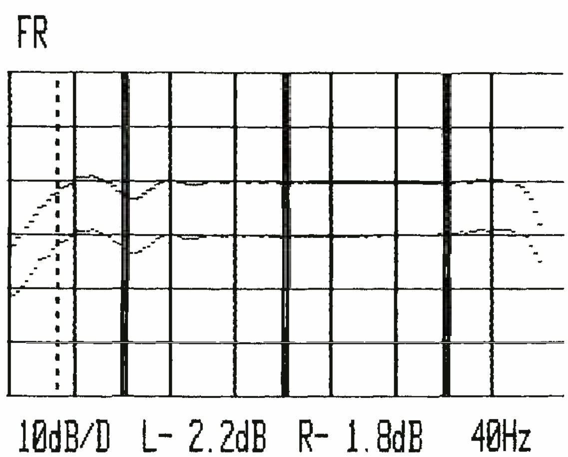
The tape transport design used in the MX-5050-MK III-8 uses two 6-pole induction motors for the tape reels and a direct-drive DC servo motor for the capstan. A pitch control is available for adjusting tape speed over a 7 percent range. In addition to the editing controls, a tape splicing block is mounted on the head cover for easy editing, cutting and tape splicing. Momentary-contact pushbutton switches on the transport are used to select record, play, stop, rewind, fast forward, and edit modes. All of these modes except the edit function can be controlled by an optionally available remote control unit, Model CB-110.

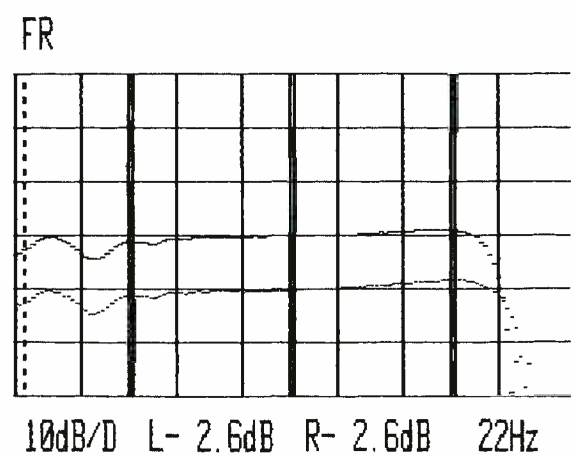
There are several other operating features worth mentioning. Available sound applications include overdubbing of sound with sound, sound on sound, etc., which can be carried out by the punch-in and punch-out functions of the transport controls. In addition to bias, level, and equalization—which can be optimally adjusted for each channel by means of rear-panel screwdriver controls—low frequency compensation controls are also provided for accurate adjustment and alignment of low-frequency characteristics to correspond with the particular tape being used. In order to make multi-channel recording even easier, the monitor programs of record-ready channels are automatically switched to input during fast-forward, rewind, or stop modes. This function can be selected with the standby switch on the rear panel. The Otari MX-5050-MKIII-8 is designed to operate on a table top, with reels oriented horizontally rather than vertically. The eight VU meters are conveniently positioned on a raised header at the rear of the instrument, placing the meters at eye level for easy viewing.

Controls and Switches: The rear panel of the recorder is equipped with two-position input and output level switches for each channel. These select -8 dBm or +4 dBm levels. XLR input and output connectors, a ground terminal, the remote control connector and a power line cord socket are also located on the rear panel, as are eight banks of fourteen small holes per channel, arranged in vertical rows. These provide access to screwdriver controls for adjusting such important operating parameters as peak level meter settings, VU meter levels, repro level, low frequency compensation, high and low frequency playback EQ, SEL/REP level, input level, record level, high and low frequency record EQ, input SRL (Standard Reference Level), internal oscillator level and, of course, record bias level. Of all these adjustments, those having to do with the tape used were factory set for Scotch 226 tape and therefore all of the definitive tests and measurements made on our sample were done using that type of 3M ½-inch tape.

The front and top surfaces of this tape recorder are also loaded with controls and useful features. In addition to incorporating the light-touch transport controls already listed, the top surface houses the power on/off switch, speed selector and reel-size selector buttons, the variable pitch control, the edit switch, memory and counter reset switches and the real-time digital tape indicator which displays elapsed time (negative as well as positive) in hours, minutes and seconds. The header-like back panel at the rear of the instrument houses the eight VU meters (each with its peak-flashing LED), individual channel input level controls, and individual SRL pushbutton switches which bypass the channel level control and select the internally preset input level. When inputting reference level, the VU meter will indicate 0 VU while rated output level will appear at the line output terminals.

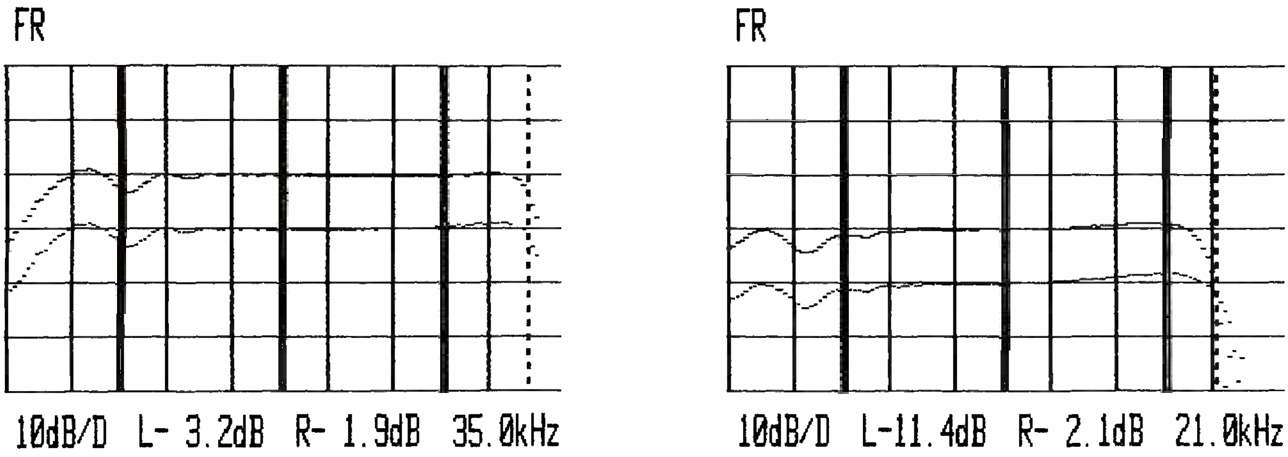
Remaining controls and switches are found on the forward front panel of the deck. There are individual ready/safe switches for each channel, individual switches that select input, SEL/REP or "repro" (tape playback) to be fed to the line outputs, a phone jack





(A) **(B)**

Fig. 1: Otari MX-5050-MKIII-8: Frequency Response Plots at 15 ips (A) and 7½ ips (B) using Scotch 226 tape. Cursor is set to show approximately -2 dB low frequency cut-off point.



(A) **(B)**

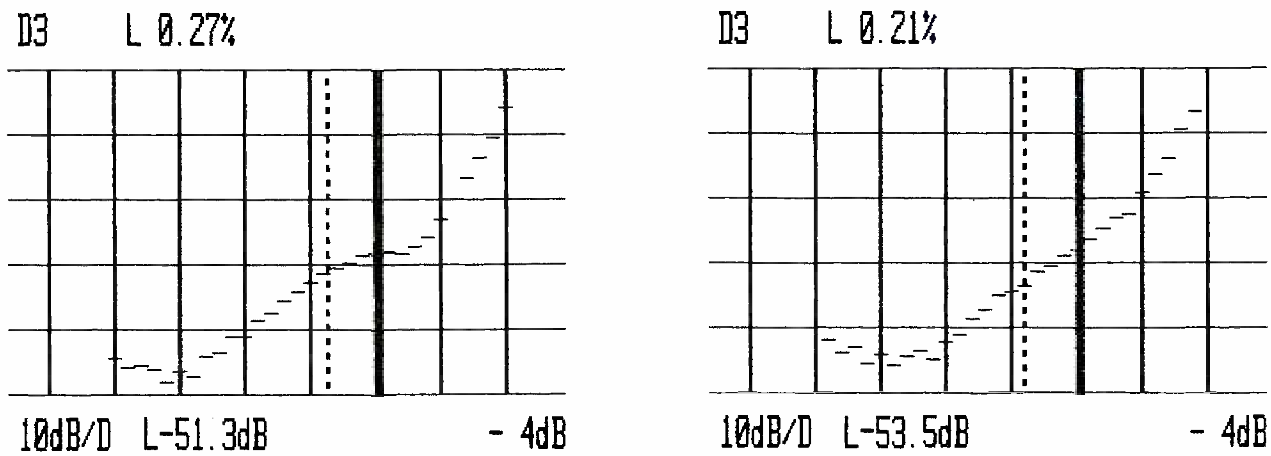
Fig. 2: Otari MX-5050-MKIII-8: Same as Fig. 1, except cursor is set to show approximately -2 dB cut-off point at high frequency end of plots.

and phone level control, individual channel buttons which determine which outputs shall be fed to the phone jack, a test oscillator selector switch, and an external oscillator connection jack. A vertically-oriented bank of five more pushbutton switches takes care of such additional functions as selecting external remote control; sending all input signals to the line-out connectors, phone jacks, VU meters and peak indicators; activating the SEL/REP function for all channels at once; sending the signals picked up by all channels of the reproduce head to the line-out connectors, phone jack, VU meters and peak indicators and, finally, permitting individual channels of input, SEL/REP or repro to be selected by the monitor selection toggle switches already mentioned.

Test Results: For all its sophistication and elaborate number of controls, the purpose of this or any other multi-track tape recorder is to make high-quality studio recordings effectively. Our laboratory performance tests were designed to show how well it accomplishes this fundamental task on a technical level. Elsewhere in this issue, you will find a hands-on evaluation of how effectively the Otari MX-5050-MK-III-8 does its job in an actual studio environment.

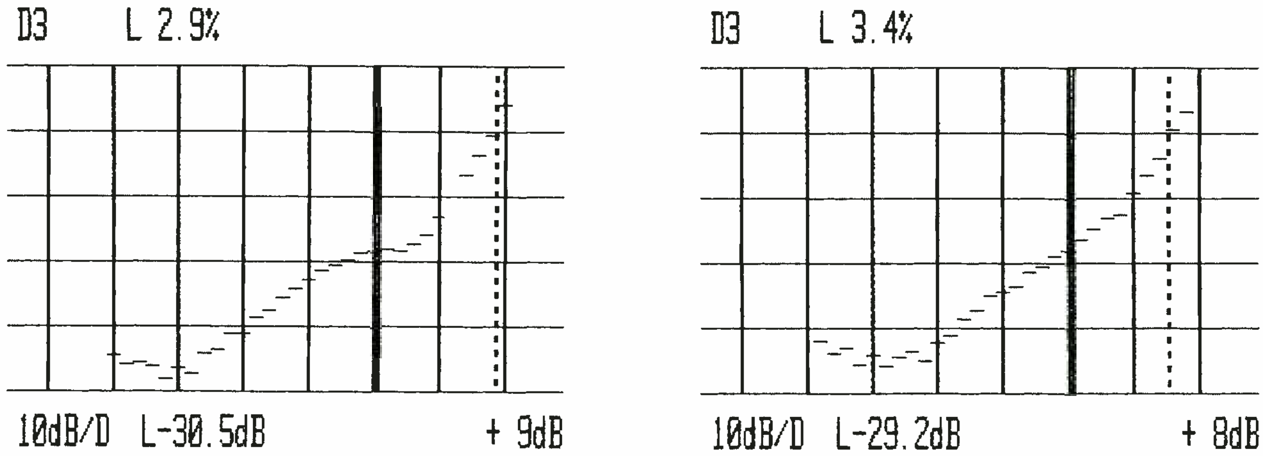
As mentioned earlier, the owner's manual supplied with the Otari deck recommends the use of Scotch 226 tape, and this is used at the factory to calibrate this machine. Since we wanted to see just how accurately that calibration had been done, we decided not to touch any of those rear panel controls at all, but simply to use Scotch 226 tape and let the measurements fall where they would. Before detailing the specific results, we must say that we were delighted with the accuracy of alignment that had been done by the people at Otari. As calibrated, Scotch 226 and the Otari MX-5050-MKIII-8 were "made for each other."

The most significant measured results of our lab tests are summarized in our usual VITAL STATISTICS table at the conclusion of this report. *Figure 1* shows plots of record/play frequency response for both operating speeds. In *Figures 1A* and *B*, record levels



(A) **(B)**

Fig. 3: Otari MX-5050-MKIII-8: Third order distortion vs. record level using Scotch 226 tape at 15 ips (A) and 7½ ips (B). Cursor is set to 0 VU (250 nWb/m) in each case (see text).



(A) **(B)**

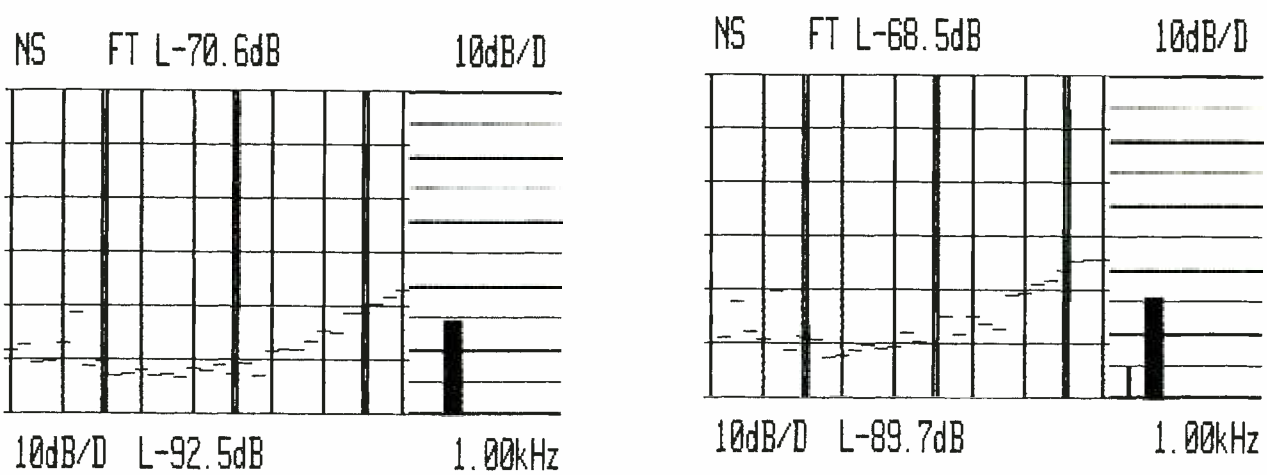
Fig. 4: Otari MX-5050-MKIII-8: Same as Fig. 3, except cursor set to read level at which 3rd order distortion is approximately 3%. Add 4 dB to readings shown to obtain max. Record level re: 250 nWb/m.

were at 0 VU (which in this case corresponds to a magnetization level of 250 nWb/m) and -10 dB). At both speeds a fair amount of head-contour effect is noted at the low end, but if we disregard this, response extends down to 40 Hz at 15 ips (for a -2 dB roll-off), and down to around 22 Hz at the slower tape speed.

Figures 2A and *B* are identical to the plots of *Figures 1A* and *B* except that the dotted-line "cursor" has been moved over to show the high-frequency at which roll-off begins. Note that at the higher tape speed, high end extends to almost the same frequency at the 0 dB level as it does for the -10 dB record level. At the lower speed, however, response is down 11.4 dB at 21 kHz for the "0 dB level" plot (designated as "L"), while it is only down 2.1 dB at that frequency for the -10 dB record level test.

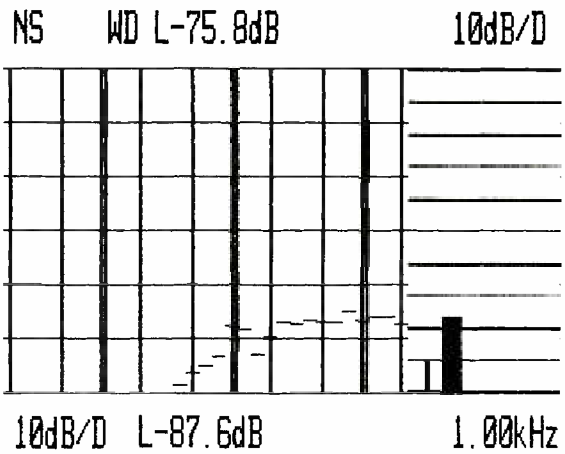
Figure 3 illustrates a plot of third-order distortion versus record level, but before examining the results, a word of explanation is in order. Normally, our Sound Technology Tape Recorder Test Set automatically begins plotting this test at a +10 dB level (referred to the 0 dB reference level which we always set at the beginning of any test series). In the case of this Otari deck, +10 dB was not a high enough level to cause a 3 percent third-order distortion (the headroom is that good). As a result, we had to deliberately increase our input signal level by about 4 dB. Thus, the double vertical line which normally corresponds to a 0 dB reference level is actually at +4 dB in the graphs of *Figures 3* and *4*. For this reason, to read actual 0 VU third-order distortion, we moved the cursor on the graph to the -4 dB point and read 0.27 percent at 15 ips and 0.21 percent third-order distortion at the 7½ ips speed.

Similarly, in *Figure 4*, when we read a +9 dB in *Figure 4A* and a +8 dB in *Figure 4B* for the record level required to produce approximately 3 percent third-order distortion during playback, we really have to add 4 dB to those readings. Thus, headroom (or mid-frequency MOL) at 15 ips was actually +13 dB (9 dB

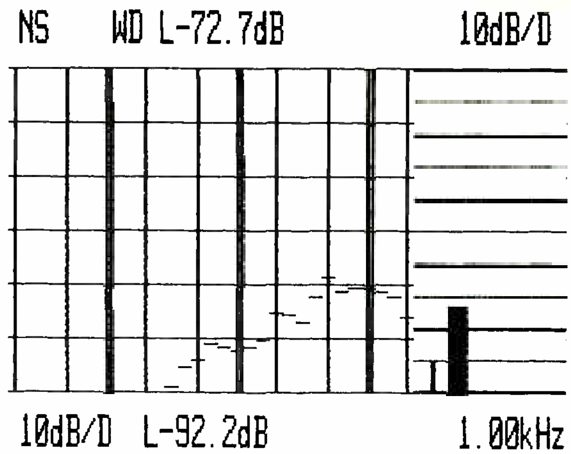


(A) **(B)**

Fig. 5: Otari MX-5050-MKIII-8: S/N analysis (unweighted) at 15 ips (A) and 7½ ips (B), using Scotch 226 tape.



(A)



(B)

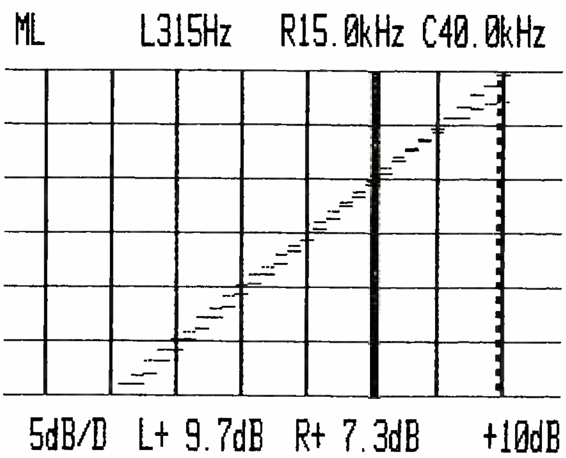
Fig. 6: Otari MX-5050-MKIII-8: Signal-to-noise analysis as in Fig. 5, except using NAB weighting curve.

plus the extra 4 dB by which we increased reference level above 0 VU), while at 7½ ips it was 12 dB. Both of these figures correspond nicely with the levels at which the peak indicators on the VU meters have been set to flash. For the signal-to-noise analyses which are represented graphically in *Figures 5* and *6*, we further increased the test set reference levels to these higher record levels (+13 and +12 dB) so that the S/N readings would be referenced to approximately 1040 nWb/m, or the 3 percent third-order distortion points, in accordance with common practice.

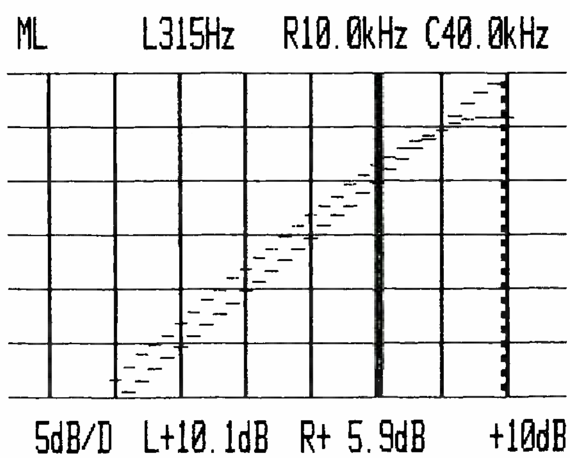
On the basis of that reference level, unweighted S/N at the 15 ips speed measured 70.6 dB, while for the 7½ ips speed the result was a S/N of 68.5 dB. Both figures were substantially better than those claimed by Otari in their published specs (see *Figures 5A* and *B*). The same held true when we used NAB weighting. Under those conditions, as shown in *Figure 6*, we obtained a S/N reading of 75.8 dB at the higher tape speed, and 72.7 dB at the 7½ ips speed.

Figures 7A and *B* represent linearity, or input versus output plots at two frequencies: 315 Hz and 15 kHz in the case of *Figure 7A*; 315 Hz and 10 kHz in the case of *Figure 7B*. In *Figure 7A*, mid-frequency linearity was very nearly perfect all the way out to +10 dB record level (9.7 dB output during playback for a +10 dB input). At that high speed, linearity for a 15 kHz signal was also amazingly good, showing only a slight degree of tape saturation at the +10 dB record level (7.3 dB output of the 15 kHz signal during playback for a +10 dB record level input).

As you might expect, results were not anywhere near as good when tape speed was reduced to 7½ ips. In fact, we had to lower the high-frequency test signal to get a meaningful reading. In *Figure 7B*, we see that mid-frequency (315 Hz) is as good as, if not better than, it was at the higher tape speed, but now even a 10 kHz high-frequency signal is fast approaching saturation when the record level is increased to a +10 dB. Output now reads 5.9 dB for an input level of +10 dB, or a non-



(A)



(B)

Fig. 7: Otari MX-5050-MKIII-8: Linearity (MOL) analysis at 15 ips, using test signals at 315 Hz (A) and at 7½ ips using test signals at 315 Hz and 10 kHz (B). Reference tape used was Scotch 226.

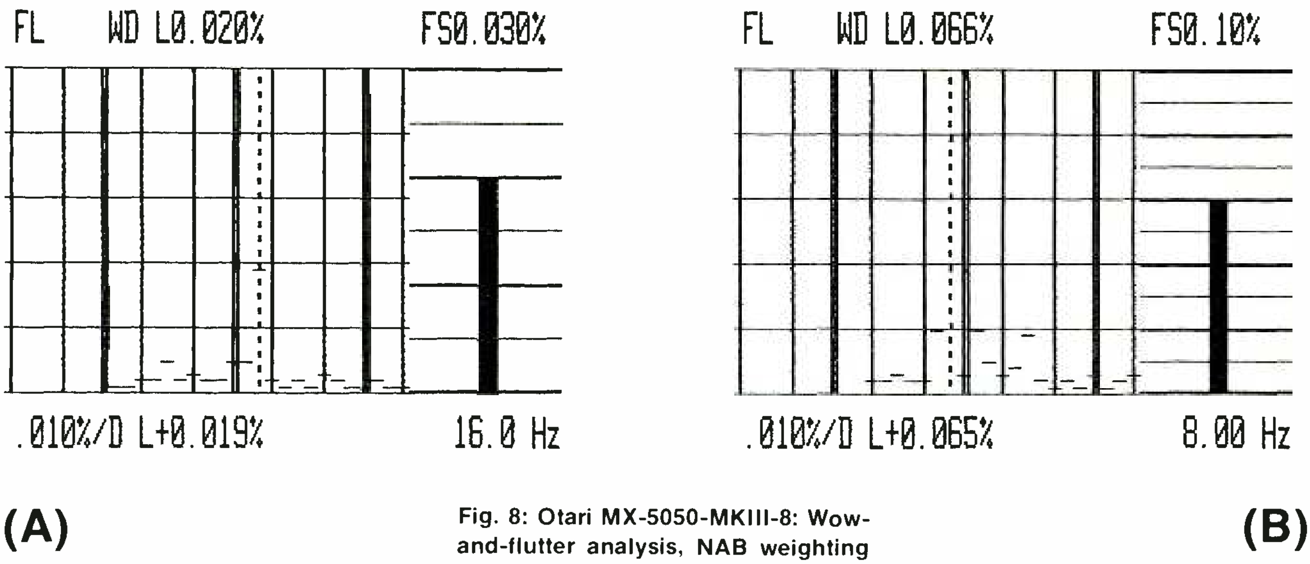


Fig. 8: Otari MX-5050-MKIII-8: Wow-and-flutter analysis, NAB weighting at 15 ips (A) and 7½ (B).

linearity of 4.1 dB.

The graphs of *Figure 8* are analyses of wow-and-flutter at the two operating tape speeds. This type of graphic analysis is extremely useful in pinpointing specific tape transport problems that cause wow-and-flutter. In *Figure 8A* we see that total NAB-weighted wow-and-flutter measured a very fine 0.02 percent. However, notice that the major component of this wow-and-flutter occurs at a most unusual frequency of 16 Hz (the graph plots from 200 Hz down to 0.5 Hz). This is most atypical, and possibly suggests that some moving part that is rotating at around 960 rpm is slightly deformed. Perhaps this may even have occurred in shipment. Note that the contribution of this one flutter component is 0.019 percent (the number at the bottom of the graph) or almost the entire total, which is only slightly higher, at 0.020 percent.

Confirmation of this deformity somewhere in the tape transport system of this recorder was obtained when we repeated the measurement of wow-and-flutter at the 7½ ips tape speed. Now the chief offending frequency of wow occurred at 8 Hz, exactly half of 16 Hz. Note, too, that the contribution of this offending frequency again almost equals the total wow-and-flutter reading: 0.065 percent as against 0.066 percent. We suspect that if this problem had not been found in this deck, the wow-and-flutter figures would certainly have been lower for the 7½ ips speed and might even have been a bit lower for the already excellent figure obtained at the higher tape speed.

In order to illustrate the importance of proper calibration of this or any other professional tape deck for the tape with which it is to be used, we decided to conduct an additional experiment. We switched to

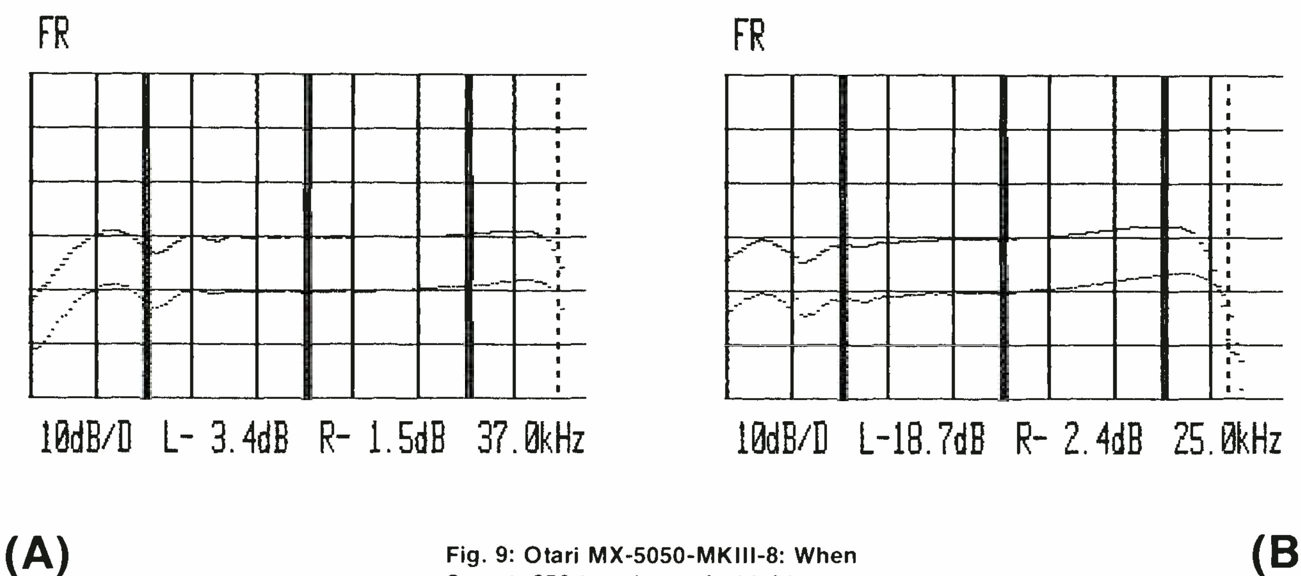


Fig. 9: Otari MX-5050-MKIII-8: When Scotch 250 tape is used with bias set for Scotch 226, high-end rise occurs at both 15 ips (A) and 7½ ips (B) indicating need for higher bias level.

Scotch Type 250 recording tape (in itself, an excellent high-quality tape formulation that many professionals use) and ran a couple of frequency response curves without altering any of the operating parameters (bias, EQ, low-frequency compensation, etc.) of the Otari MX-5050-MKIII-8. The results are shown in the graphs of *Figure 9*. At the higher tape speed *(Figure 9A)*, deviation from flattest possible response was not too severe, although a slightly rising high end is clearly in evidence. Switching to the slower tape speed, however *(Figure 9B)*, we see a fairly lopsided response curve, with a sharply rising (overly brilliant sounding) high-end and a seriously attenuated low-end that could obviously use readjustment of low-frequency compensation.

If the results in *Figure 9* aren't enough to convince you of the importance of matching and calibrating a fine deck such as this Otari MX 5050 MKIII-8 to the tape with which it's to be used, you might want to take a look at *Figure 10*. Here, we reverted to the recommended Scotch 226 tape once again, but instead of leaving the bias control set where it has been, we rotated it first all the way to one extreme and then to the other, in each case plotting the resulting record/play response curve. Enough said?

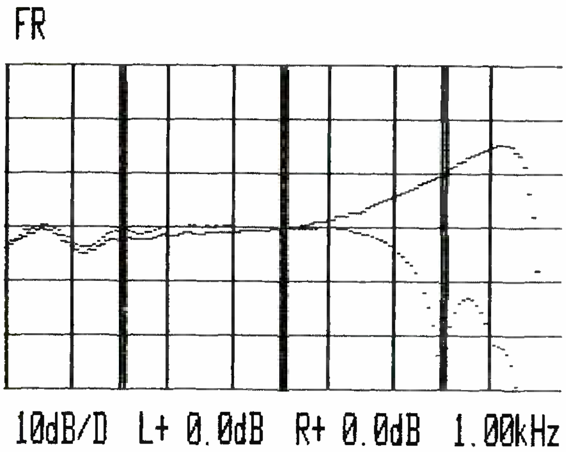


Fig. 10: Otari MX-5050-MKIII-8: Offers very wide range of bias adjustment. Here, response curves were plotted for Scotch 250 tape with bias adjusted arbitrarily for minimum and maximum settings.

Individual Comment: As you have surely gathered by now, I was very favorably impressed by both the mechanical and electrical aspects of this excellent 8-track deck. Despite its great number of controls and switches, everything is so logically laid out that it took me only a short time to become thoroughly familiar with the operation of the recorder. The accompanying operating manual is well-written, too, and includes complete trouble-shooting and servicing procedures as well as clear schematic diagrams and board layout diagrams. While I can speak about the measured performance and relative ease of use of the recorder, I defer to Michael Tapes, who put the machine through its paces after I finished my lab evaluation of it. His hands-on report follows.

OTARI MX-5050-MK III-8 REEL-TO-REEL RECORDER: Vital Statistics

SPECIFICATIONS	MANUFACTURER'S SPECS	LAB MEASUREMENT
Tape width & tracks	½-inch/8 tracks	Confirmed
Tape Speeds	15 and 7½ ips	Confirmed
Maximum reel size	10½ inch	Confirmed
Heads	3 8-track in-line	Confirmed
Rewind Time (2500′)	100 seconds	82 seconds
Line Input Level	+4 dBm/-8 dBm, switchable	Confirmed
Line Output Level	+4 dBm/-8 dBm, selectable	Confirmed
Headphone level	-19 dBm	Confirmed
Equalization	NAB or IEC for 7½ & 15 ips	Confirmed
Frequency Response (Rec/Play)		
15 ips	40 Hz to 25 kHz, +/-2 dB	40 Hz to 35 kHz
7½ ips	20 Hz to 20 kHz, +/-2 dB	22 Hz to 21 kHz
Signal-to-Noise Ratio:		
NAB Wtd, 15/7½ ips	70/70 dB	75.8/72.7 dB
NAB Unweighted, 15/7½ ips	66/66 dB	70.6/68.5 dB
IEC Weighted, 15/7½ ips	70/70 dB	N.A./N.A.
IEC Unweighted, 15/7½ ips	66/66 dB	N.A./N.A.
Crosstalk	Better than 55 dB	More than 60 dB
Wow-and-flutter, NAB W'td, 15/7½ ips	0.05%/0.06%	0.02/0.066%
Distortion, 1 kHz, at 250 nWb/m	Less than 0.5%	0.27%
Erase Efficiency	Greater than 70 dB	Confirmed
Test oscillator frequencies	1 kHz and 10 kHz	Confirmed
Bias and erase frequency	200 kHz	Confirmed
Peak Indicator Level	1040 nWb/m	800 nWb/m
Dimensions	17.3″ w. x 26.6″ d. x 17.3″ h.	Confirmed
Weight	77 lbs.	Confirmed
Power Requirements	100, 117, 220, 240 V, 50/60 Hz AC	
Power Consumption	150 watts	135 watts

Suggested List Price: $5295.00.

Otari MX-5050-MKIII-8

Michael Tapes

I was excited at the prospect of doing a "hands-on" review of the latest Otari half-inch 8-track tape recorder. Over the years, in addition to working in traditional (one-inch, eight-track) recording environments, I have had considerable involvement with the half-inch eight-track format. In fact, I had been very involved with the original machines that boasted the then-new format, and served as a consultant in the design of the second-generation machines. I have also had quite a number of hours using Otari's now discontinued half-inch eight-track recorder, the "8D."

When *MR&M* editor John Woram asked me to do a "Hands-On Report" on the Otari MX-5050-MKIII-8, it fit perfectly into my plans. I had just convinced myself to get back into my role as a musician (I hadn't played the drums in about six years, except for digital drums—thanks to the wizardry of Roger Linn). I've just finished the construction of a new music room which is to serve at different times as a studio, a rehearsal hall, and a test lab for new and existing Sound Workshop console designs. The "studio" is presently outfitted with a 20 x 16 Sound Workshop Logex Mixing Console with assorted peripheral gear, the now-resident full Yamaha Recording Series Drum Kit, the Linn LM-1 Drum Computer, a Fender Bass and various keyboards including an exciting new design from Digital Keyboards called "Synergy." Most of my musician friends don't need much of an excuse to drop what they are doing to get involved in a recording project, so I figured I had the proper ingredients to give the Otari a good workout.

I received the Otari after it had undergone a series of lab tests by Len Feldman. Because of this, I planned to limit the scope of this report to my own in-use experience. Since I knew that Len would give a full description of the features of the machine, as well as a total overview of its electrical performance, I planned not to worry abut every small detail, but to use the machine as if I had just purchased it. Since I have a tendency to review everything I purchase anyway, this

approach seemed perfect.

The machine comes in one large carton and the packaging design is excellent. It is definitely the kind of box that should be saved, in case the machine ever has to be shipped to another location. Speaking of shipping to other locations, the MX-5050 is not a small machine. Consistent with its totally professional design, it wants to be installed and enjoyed. If it is being used in a home environment, it will not be easily moved to the studio to transfer the tracks to a larger format machine—although it is not impossible.

Interfacing

After unpacking, I set up the machine and interfaced it to the Logex console. Checking out the rear panel, it was apparent that the interface would be simple and straightforward. The eight inputs and eight outputs terminate in XLR-type connectors. Both the inputs and outputs are unbalanced, but are fully compatible with both balanced and unbalanced circuits. The rear panel also contains an impressive array of alignment controls. Among these controls are output and input level switches marked H and L (High and Low). I ensured that all switches were set to High for proper interface with the +4 dBv levels of the console. (The L position of the input and output switches sets the levels for −8 dBv).

All of the setup and my first dealings with the machine were accomplished without looking at the instruction manual. While it is true that I have much experience with tape machines, and I have used other Otari products, I feel that my ability to proceed unaided was mostly due to the logical and straightforward layout and function of this machine. Both novice and professional engineers will find this a refreshing machine to use. I used the instruction manual as a technical reference, and in this fashion it served me well. It is not a step-by-step operation guide; it is factual in nature and provides a wealth of data and drawings, but it will not hold you by the hand and teach you the art of multi-track recording. This once again fits into Otari's professional approach.

Once the machine was hooked up, I wandered around the front panel. Like the rear, it's logical, complete and straightforward. To check out my installation, I turned the TEST OSC (Oscillator) switch to 1K, and pressed the ALL INPUT button. The input LEDs assured me that all the channels were in the Input mode. A glance at the meters indicated all channels at "0," while a look at the console indicated the same. The rest of the installation check went as smoothly.

After a more detailed examination of the front panel, I began to feel right at home with the Otari machine. There are master buttons for switching all eight channels into Input, Sync, or Reproduce, and there are also provisions for allowing each channel to be switched individually into these three modes.

Regardless of how the channels are controlled, each channel has three LEDs to indicate which mode it is in. Additionally, there is a master button that diverts channel control to the optional remote control unit (model CB-110). Even when under this EXTERNAL CONTROL, the LEDs on the machine indicate channel status.

For each of the eight channels there is also a ready/safe switch with an associated LED. When a channel is put into record ready, the LED flashes to indicate the ready status. When actually recording, that channel's LED goes into a full-on state. This in combination with the other LED status indicators provides a vital and very graphic indication of exactly what the machine is up to at all times.

Of course, as with any professional tape recorder, the proof of performance comes when the record light comes on and the musicians start to play. In essence, a tape recorder has two main functions to perform: It must make a great sounding high quality recording, and it must be easy to use. In the case of a professional multi-track tape machine, easy to use translates into fast, flexible and friendly. This operator ease is equally important in a professional studio, a home studio, or a production facility.

The MX-5050 in Use

To most fully evaluate the operation of the Otari, I used it on several different sessions, where I served as recording engineer, musician, or both. The machine felt very comfortable and operating it came second nature with few exceptions. The transport was responsive to all commands and the machine has microprocessor control of its motion sensing and full dynamic braking systems. Tape handling seemed smooth and safe. The microprocessor also controls the digital tape timer. This LED readout indicates hours, minutes, and seconds in both positive and negative increments. A problem with the readout stems from its physical position. The transport on the Otari is in a horizontal plane and the function control panel is vertical at the front of the machine. In setting up the machine I positioned it so that the front function panel was at eye level, with the transport controls easily within reach on the top. However, this kept me from seeing the time readout which is mounted flush with the transport; it was a bit inconvenient. The truth is that even some of the most expensive recorders in the world suffer from this same syndrome. I have heard that there are prisms available (probably from Edmund Scientific) that would allow viewing the timer from the front...Sounds like an inexpensive solution.

Tied in with the timer is a search-to-zero function. It seems like this is a cross between the consumer-oriented memory rewind function and the professional search-to-zero. It functions like a search in that, regardless of the tape position, when MEMORY is pressed the transport goes into whatever mode it must to return to zero. The problem is that when it reaches zero, it simply issues a stop command. If speed had been attained by this time, the transport could overshoot zero by as much as 35 seconds at 15 IPs. About one-half second after the tape stops, MEMORY can be pressed again, and now the tape will stop within a few seconds of zero. Two other inconveniences exist with the search-to-zero function: 1) It cannot be engaged from the record mode. This means that after a take one must press STOP and then MEMORY to return to the beginning of the song. 2) You cannot make the machine search to zero and then play without waiting for the machine to stop first. It would be nice while recording to be able to stop the recording just by pressing MEMORY, and then lean back while the machine shuttles back and plays that magic take.

On the other side of the ledger, all of the channel function controls are perfect. During recording, overdubbing, or punch-ins, the "ready/record" channels automatically switch from sync to input (and vice versa) just as they should. There is absolutely never a time when more than one finger is needed. Critical communication with the musicians is maintained during all normal machine functions. (Record/ready channels did drop out of input momentarily when stop was engaged. This is a minor inconvenience, and the factory tells me that it is being looked into.) The layout of the function panel, and its switches and LEDs, proved to be exceptional in use.

The sonic performance of the Otari was excellent. I didn't do any electrical measurements until after I had done sonic evaluations. My electrical measurements confirmed what my ears had heard. Response was extended and smooth. This is especially important when doing sound-on-sound recordings (bouncing tracks). In addition, signal-to-noise and cross talk were better than expected, given the half-inch 8-track format. Is it as good as a one-inch 8-track machine? No.

Where it falls short is in its ability to handle high-frequency dynamic transients. But, of course, that is where all analog tape recorders fall off. While I didn't do an A/B with a one-inch 8-track machine, it is apparent that a one-inch would offer more overall dynamic range and transparency than a half-inch machine. But this comes as no surprise. Certainly the Otari half-inch machine used with an overall consciousness of the limitations of its format could outperform a one-inch machine used with a disregard of conservative recording techniques.

Summary

My overall impression of the Otari MX-5050-MKIII-8 is that it is a fully professional and competent tape machine in performance and function. It performs efficiently and effectively and is easy to adapt to.

My only criticism is with the timer and related search-to-zero functions. Since my initial impressions, two things have occurred. I have been sent the optional CB-110 transport/function remote control, and I have spoken to the factory about my disappointment with the search-to-zero functions.

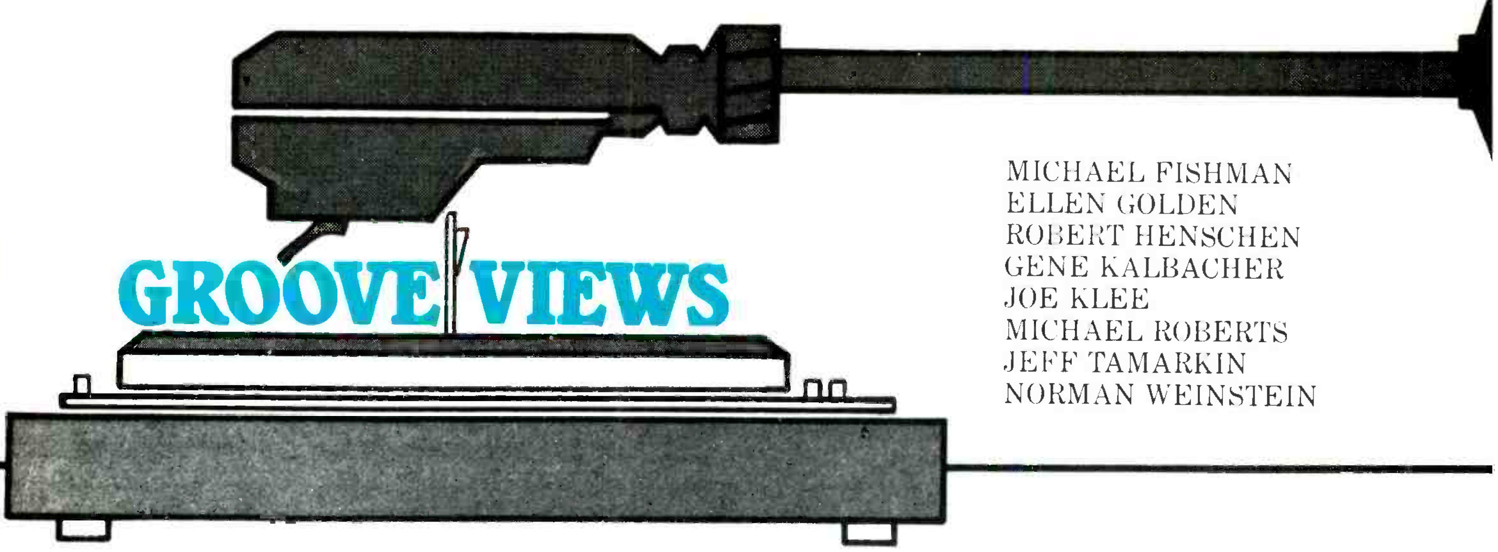
The remote is just what I needed. On one small panel are the channel function controls, the transport controls, *and* the timer readout. This allows the machine to be located remotely, and all relevant functions to be performed from the remote panel. This unit adds to the convenience of the machine, and alleviates my criticism of the location of the timer.

After speaking with the factory about the search-to-zero functions, I was informed that soon after this is in print the optional auto-locator will be available. I was given a run downdown of the features of this unit and it will not only accurately search and stop precisely at zero, but it also has eight memories for various search cue-point functions, and a host of other functions that I haven't even seen on 24-track auto-locators. These include head and tail guard memories, and complete offset of existing cue-point when zero is reset. I hope to review this unit completely when it becomes available.

Since I knew that Len Feldman was writing a lab report of the Otari machine, I have assumed that he has at least described all of the other features included in this machine. These include convenient editing provisions, a simple but handy headphone mixer and amplifier, VU and peak metering, variable speed operation, internal test oscillator, tape lifter defeat, calibrated and variable line input level controls, and extensive alignment capability.

Just to familiarize myself with the machine and to confirm my sonic judgements, I did briefly go through an alignment of one channel. As long as you allow access to the back of the machine, the alignment facilities are excellent. I especially appreciated the EXT OSC jack which allowed me to feed all eight tracks without having to involve the console. This is especially convenient in the studio where the console can be used for other functions while the machine is being aligned. The phone's mixer/amp also comes in handy here. Of course these features and the built-in oscillator make the fast system check very simple.

Based on my in-use evaluation of the Otari MKIII-8, I can recommend it highly as a professional 8-track recorder applicable in both music recording and production environments. Its performance both sonically and operationally is excellent. While I can only speculate on its long-term reliability, based on its construction and competent alignment provisions, as well as my experience with other MX-5050 machines, it appears that it will maintain its high performance standards over the years. For those considering moving up to an 8-track machine, the Otari can be considered a big step in the right direction. In many applications, its initial cost, plus the savings on tape, may make it an attractive choice where only 1-inch machines may have been considered in the past.



GROOVE VIEWS

MICHAEL FISHMAN
ELLEN GOLDEN
ROBERT HENSCHEN
GENE KALBACHER
JOE KLEE
MICHAEL ROBERTS
JEFF TAMARKIN
NORMAN WEINSTEIN

POPULAR

JONI MITCHELL: *Wild Things Run Fast.* [Joni Mitchell, producer; recorded at A&M Studios by Henry Lewy and Skip Cottrell, assisted by Clyde Kaplan, with the exception of "Be Cool" which was recorded at Devonshire Studios, engineered by Jerry Hudgins; mixed at Paramount Studios by Larry Hirsch, Larry Klein and Joni Mitchell, assisted by Chase Williams; originally mastered by John Golden at K-Disc.] Geffen GHS 2019.

Performance: **Would brilliant be pushing it?**
Recording: **Quality crystal**

There could not have been a more perfect title for Joni Mitchell's latest album than *Wild Things Run Fast.* Having abandoned what we could loosely call "serious jazz excursions," Mitchell has flexed a tighter pop muscle and one gets the feeling that this is the territory where she can do the most good for herself—not to mention those of us who still chase stars to the beat of our hearts. Not only is Joni Mitchell wild, but she's running fast 'cause she's in love. Funny how love of life and lust can sometimes make for *great*, exhilarating vinyl.

Yet no one would get the chance to feel any of Joni's massive energy if she wasn't such a damn good producer. Besides writing every song but Jerry Leiber and Mike Stoller's "(You're So Square) Baby, I Don't Care" and Hy Zaret's inserted "Unchained Melody" lines into her own "Chinese Cafe," Joni has produced an album that cuts with its edges. (To be fair: Rhythm arrangements on "Wild Things Run Fast" and "You're So Square" are Larry Klein and Vinnie Colaiuta; the arrangement on "Be Cool" is by Don Alias.) We wouldn't feel so good listening to this record if we couldn't hear every little vocal expression, every rhythmic choice, every time signature change, all the highs, the gentle lows, etc. Joni Mitchell is all about starting up in one vocal and musical direction, getting us moving (feet tapping, gut wrenching, whatever), then zipping around for a line or two with those golden vocal chords of hers. Just because this is an uptemp LP doesn't mean that she hasn't taken that suspiciously improvised musical style of hers to the nth degree. This lady is a pro, and anyone who gets a copy of *Wild Things Run Fast* will get a chance to *hear* (how many times do I have to emphasize that?) this master at work.

I guess *Wild Things Run Fast* will

officially be tagged a rock album, since I've seen "You're So Square" on my local bar jukebox right there in between the popular A Flock of Seagulls and Men At Work. The difference between those trendos and Mitchell's single is that her work comes from the belly—and you can tell. Her's is American rock about a guy who prefers movies and necking in the park to parties and sleazy bands. She wraps her southern inflections with melting instruments and proclamations of "Baby, I Don't Care." Elvis Presley and Buddy Holly would be proud; they both had their own versions in the good old days.

Although there's the threatening drum beat of "You Dream Flat Tires," Joni still seems a bit disenchanted with backbeats in general. "Chinese Cafe" proves this as the album's opener, when a brazzy cymbal is given front and center treatment along with an acoustic piano, a synthesizer, Joni's overdubbed backup vocaled chorus, and the insertion of "Unchained Melody"—the 1955 hit whose romantic notions are very much akin to Joni's new starry ideas. As for the drums, they are simply a jazzy blur of color. It works. "Moon At The Window" features brushes on the snare, admittedly a constant rhythm, but an odd sound for a compositional base. But again, it works, and that's what counts.

With the Caribbean rhythms leading the way in the very upbeat "Solid Love," all the instruments tend to stay in line and veer less from a format considered palatable to the masses. "Solid Love" may be Joni's tribute to the Police, a band she had hoped to work with on this record, but commitments—their's, not her's—prevented that union. Nevertheless, *Wild Things Run Fast* does feature an impressive array of guest musicians, vocalists and pals. Lionel Richie sings a few lines in the hard rock—piercing electric guitar and all—"You Dream Flat Tires," while the whisper chorus of "Be Cool" features Kenny Rankin (among others). James Taylor says a vocal hello and Larry Carlton and Chase Williams both add their musical expertise as well.

Despite the cast of thousands, this is still Joni Mitchell's musical movie. To find the record speaking from beginning to end about love, love, love is only half the film. Without her compassionate song constructions and verbal melodies, who would care about the plot? EZG

STACY LATTISAW: *Sneakin' Out.* [Narada Michael Walden, producer; David Frazer, Leslie Ann Jones and Tom McCarthy engineers; recorded at The Automatt, San Francisco, CA, Room 10 Recording Studios, Washington, D.C. and The Record Plant, Los Angeles, CA, 1982.] Cotillion 90002-1.

Performance: **Motown**
Recording: **Likewise, I'm sure. Pardon the California studios.**

Clap-clap, finger snap. Shuffle-kick, point toe and turn around. Yes, yes, the style is unmistakably that Motown sound of the five brothers Jackson. Right down to, and including, the memorable "Hey There Lonely Girl,..." pardon me just a moment. (Okay, side one, cut 5, yeah, huh!?!) Strike that.

You know, I could have sworn (though not in print) that that was Michael Jackson and the Jackson Five. Wow! I mean, like, was I surprised or what man. It wasn't, nor is it, the J-5, it's a relative newcomer named Stacy Lattisaw.

Sixty-six percent (or two thirds) of the songs on this LP are written by Mr. Narada Michael Walden. Mr. Walden has also produced this album under Perfection Light Productions. Then to find that he is credited with being the drummer, keyboardist and percussionist as well, should not surprise me, considering the sameness of the songs. Although the album is well produced, via excellent recording facilities, the greatest shortcoming must be attributed to

Stacy Lattisaw

the apparently one-dimensional approach of Mr. Walden's. Don't get me wrong, he has worked hard—maybe too hard. And let's face it, using a successfully proven formula reminiscent of the Jackson Five could vault Stacy Lattisaw to national prominence. If she just wasn't made to sound like Michael Jackson's clone.

Stacy Lattisaw sings lead vocals on every tune. She sounds young. And, unfortunately for her, the nostalgia that arises from her uncanny aural likeness to Michael J. prohibits her from defining a place for herself in the minds of music lovers.

The remake of "Hey There Lonely

Boy" isn't the only thievery from the past. Cleverly disguised under the cloak of the video arcade scenario, and highlighted by some creative synthesizer trickery of Patrick Cowley and Martian impresario by co-writer Jeff Cohen, is the "Attack of the Name Game." Music triviaites will will instinctively recall the "bananna-fananna-bo-banna" of this sixties neo-classic.

There were, and are, many happy hours of listening on this album. It's sophisticated bubblegum. Unlike gooey, globby, nauseous regular bubblegum music. Mr. Walden has created a significant undertoning of back-up artists that make this a viable listening adventure and a credit to Warner Communications and the studios where it was produced.

I would venture to say that marginal MOR air-play lies in a combined 45 release of "Hey There Lonely Boy" and "Attack of the Name Game." No hits, no big album sales, and no star making possibilities for Stacy Lattisaw, but the novelty and nostalgia of these two songs do have their appeal.

The album should also appeal to a certain segment of the (quote) record buying public (end quote), if indeed they still exist. There are those who will like the up-beat, tap-your-feet, disco-funk which is most prevalant on side one of *Sneakin' Out*. Then for those who find ballads and lilting refrains more to their liking, side two is for you.

Now how about that? An album complete with everything this discriminating cross-section might want, plus, they don't even have to flip it over! Neat, huh?

Anyway, as you listen to this album and recollect on the days when the J-5 were the hottest soul act around, please remember: this is a female, her name is Stacy Lattisaw, and she has relatives in this great land of ours who are proud as punch that one of their own has made a record—any record—even if Michael Jackson could hit the high notes better.

M R

GRACE JONES: *Living My Life.* [Chris Blackwell and Alex Sadkin, producers; Sadkin and Steven Stanley, engineers; recorded at Compass Point Studios, Nassau, Bahamas.] Island Records 90018-1.

The Passion of Form: George Russell and John Lewis

By Nat Hentoff

Ever since he startled the New York jazz scene in 1947 by writing "Cubana Be, Cubano Bop" for Dizzy Gillespie's big band, George Russell has been one of the most consistently invigorating composers in jazz. A most lucid man, off as well as on the stand, Russell is fascinated by the uses and potentiality of musical form. In fact, he has created his very own microcosm of musical structure—the Lydian Chromatic Concept of Tonal Organization. But while this sounds academic and abstract, the actual music created by Russell is decidedly, and powerfully, emotional. Indeed, few jazz composers can build to such organically exciting climaxes as Russell.

His most natural expressive context is the big band, as is vividly evident in *George Russell's New York Big Band* on the Italian Soul Note label—now available in the United States through Polygram Special Imports. There is a crackling new performance of "Cubana Be, Cubano Bop" by the Swedish Radio Jazz Orchestra, but all the rest is the Russell big band in such pieces as "Listen to the Silence," "Big City Blues," and "Living Time."

Each work is intriguing on many counts—the precise but subtle textures, the inexorable melodic development, the rhythmic imagination, and the highly judicious use of electronics. ("I only use electronics as an added color," he says. Electronics are never dominant in his work.)

The sound quality of this 1978 New York recording is as brilliantly balanced as Russell's music. All elements in his protean forms are delightfully clear.

John Lewis was also in Dizzy Gillespie's big band in 1947, and he too had a remarkable work of his premiered by Dizzy that year—"Toccato for Trumpet and Orchestra." Also like George Russell, Lewis' subsequent writing has been characterized by his passion for the discipline of form while the content of the music has been deeply lyrical and often laced with the blues.

As musical director of the Modern Jazz Quartet, for example, Lewis proved that the spontaneity of jazz could be enhanced by structuring improvisation in luminous and resilient patterns. Now, Lewis, who teaches jazz at New York's City College, has found yet another way to fuse form and spontaneity. It's the John Lewis Group, and its first album is on Finesse Records.

With Lewis on piano (and resident composer as well), the group consists of bassist Marc Johnson, guitarist Howard Collins, drummer Shelly Manne, flutist Frank Wess, and the continually imaginative violinist, Joe Kennedy Jr. As is nearly always the case in a John Lewis combo, the focus is on strong, lean swinging as well as on soaring melodic improvisation. On a number of these tracks, the sheer excitement of this collective interplay is exhilarating, and none of the performances are ever less than resourceful.

The engineering is thoroughly up to the exceptionally clean-textured interplay—and is also superbly right for that nonpareil John Lewis sound on piano. This is so clear a sound that it goes deeply to the roots of both feeling and time.

JOHN LEWIS: *Kansas City Breaks.* [Ken Glancy, producer; Don Puluse, engineer.] Finesse Records FW 38187.

GEORGE RUSSELL: *New York Band.* [George Russell, producer; Bob Clifford, engineer.] Soul Note SN 1039.

Performance: **Pleasing, all-integrative, tight**
Recording: **Crisp, dynamic, inimitable production**

Inroads to the pop mainstream have traditionally been obstructed for even the most formulaic of black musical artists. While top 40 radio embraced the Motown sound, and still nurtures all of that phenomenon's offspring, black artists as widely accepted as Michael Jackson and Stevie Wonder still cannot break the impenetrable music policy of MTV, the rock music cable network still in its infancy. That considered, the truly *progressive* black artists are even more at odds with the powers that provide access to the most visible media. However, people like James Blood Ulmer, Grandmaster Flash and Grace Jones are progressives whose burgeoning numbers of loyal fans have just started to afford them some degree of favorable notoriety in the recording arena.

Jones' *Living My Life*, as a follow-up to 1981's *Nightclubbing*, is another provocative, densely-packed anthology of urban/island funk-reggae-disco-derived songs. For all its utilization of third-world rhythms, though, the specific instrumentation and resulting textures on this recording are at once well-conceived and thoroughly refreshing. This is directly attributable to the unique sensibilities of co-producer and Island Records principal Chris Blackwell, whose contributions here are marked by rich sonic backdrops over which Jones' deep, authoritative voice careens and glides perfectly complemented.

The supporting musicians on the date, aside from individuals long familiar with each other's styles, are veterans of Blackwell's recording projects.

The technical challenges of *Living My Life* have been confronted beautifully, right down to the pressing, which has thankfully bypassed the ills routinely associated with non-audiophile domestic pressings. The mix is clean, the players cooperatively targeted towards the creation of a fitting sound tonic. Since bassist Robbie Shakespeare and drummer Sly Dunbar are the real "stars" of the band, their work is mixed louder than would be the playing of lesser studio presences. The guitars of Barry Reynolds and Mikey Chung and the keyboards of Wally Badarou are nicely woven into a supportive whole far greater than the sum of the individual contributions. The band is hot, and the recording shows it.

Musically, Grace Jones' personality is luminous and imposing. With "My Jamaican Guy," Jones extends herself so far as to describe a lover in his own island patois, but immediately backs off for an assertive, individualistic stance on the album's first single, "Nipple to the Bottle." Emotions elsewhere run the gamut between these two extremes, as with the predicaments of "Everybody Hold Still," "Cry Now Laugh Later,"

Grace Jones

and "Inspiration"—all written with Barry Reynolds and featuring his striking guitar work.

While the surprising inclusions of *Nightclubbing* were familiar material by Bill Withers and David Bowie, the only non-original on *Living My Life* is "The Apple Stretching," the Melvin Van Peebles piece from his Broadway show, *The Waltz of the Stork*. This is largely a bitter narration depicting the loneliest characters in New York at dawn; lyrically, it strengthens the album, but musically it weakens it.

Although Jones' recent recordings have included the same supporting players, and were recorded by the same producers and engineers in the same studio setting, the changes in her music are still evident; with a strong, daring compositional sense, Jones is well able to mark each successive date as a progression from the last, and a venture into what has thus far, fortunately, proved a marketable frontier. MF

CLASSIFIED ADS

Sunn Custom mixing Bd. with 32 inputs, 10 outputs, all balanced, with road case, 180pt. patch bay, and two 25pt. snakes. 3K (804) 929-0704.

EIGHTH DAY SOUND
The full line sales and service company. For all your pro Audio and Sound Reinforcement needs. For the absolute best prices and fast, reliable service on all major brands of audio or for complete Concert and Touring Sound Production, Call the Professionals. Eighth Day Sound, 2104 Superior Ave., Cleveland OH 44114. (216) 566-7303 or (216) 255-6884.

AUDIOHOUSE SELLS Valley People, Mic Mix, Ecoplate, Symetrix and EXR Exiter at the best prices. (303) 751-2268.

WANTED

Full Time **INSTRUCTORS** Part Time

FOR AUDIO RECORDING TECHNOLOGY

To teach basic and advanced theory and operation of multi-track recording studio equipment.

Audio field experience required. Teaching experience preferred.

Location—historic Greenwich Village.
Salary—attractive; commensurate with background, experience and responsibility of position.

Send detailed resume, including salary history to:

Philip Stein, Director
Institute of Audio Research
64 University Place
Greenwich Village
New York, N.Y. 10003

WANTED: McINTOSH, MARANTZ, WESTERN ELECTRIC TUBE TYPE AMPLIFIERS. PREAMPS, JBL HARTSFIELD & EV PATRICIAN SPEAKERS. (713) 728-4343, Maury Corb, 11122 Atwell, Houston, TX 77096.

FREE your copy of yesterday's prices on the industry's leading Pro Audio and Stage Lighting Equipment and Accessories. Mixers, amps, equalizers, mics, lights, dimmers, lamps, gel and much more. Check our prices before you buy. All-Pro Audio and Lighting, P.O. Box 3992, Centredale, RI 02911. Big Savings from the Smallest State.

Portable Equipment. UHER, Sony, Sennheiser, AKG, Audio-Technica. Send SSAE; Model, Quantity. Carpenter Sound, Box 1321, Meadville, PA 16335-0821.

CASSETTE DUPLICATION in Real Time, from 10 to 10,000. Highest quality, low rates. Labels, inserts and shrink wrap available. Fast turnaround. AUDIOHOUSE (303) 751-2268.

BLANK AUDIO & VIDEO CASSETTES: Direct from manufacturer: Below wholesale: Any length cassettes, four different qualities to choose from. AMPEX and AGFA MASTERTAPE: from ¼" x 2". Cassette duplication also available. BROCHURE. ANDOL AUDIO PRODUCTS INC. 4212 14th Ave. Dept. MR, Brooklyn, NY 11219. TOLL FREE #1-800-221-6578/N.Y. RES. 212-435-7322, ext. 2.

Scully M-100 16-track recorder. DRC 2846 Dewey Ave. Rochester, NY 14616 (716) 621-6270.

FREE CASSETTES—On orders of 100 or more—you receive 10 free (same excellent quality). Money back guarantee. C-30 .85, C-40 .95, C-60 1.17 DRC P.O. Box 105, Greece, NY 14515.

CLASSIFIED

- **PRODUCTS**
- **EQUIPMENT**
- **SERVICES**

Classified Rates $1.00 per word. Minimum $25.00. Copy must be received at Modern Recording & Music, 1120 Old Country Road, Plainview, NY 11803 by the 15th day of the 2nd month prior to cover date (for example, the April issue closes February 15th). Payment must accompany order. Phone numbers count as 1 word. Zip codes are free.

DISPLAY ADVERTISING $40.00 per column inch.

Advertiser's Index

R.S. #		Page #
23	Carvin	19
28	Countryman	31
13	Crown	3
No #	Delta Lab	21
16	Electro-Voice	7
19	Furman	10
14	Gold Line	4
25	Ibanez	25
34	Institite of Audio/ Video	62
35	Linear Digital	62
No #	LT Sound	35
No #	LT Sound	60
17	Omega Studio	8
32	Orban	39
12	Otari	Cov. 4
29	PAIA	35
21	Phase Linear	13
30	Polyline	36
18	QSC	9
27	Rane	28
36	Sam Ash	37
24	Sound Productions	24
15	Soundtracs	5
20	TDK	11
10	Tascam div. of Teac Corp.	Cov. 2
22	Tascam div. of Teac Corp.	17
33	Telex	43
31	Trebas	36
11	Unicord	Cov. 3
26	Unicord	27

FOR SALE: Tascam Model 5A; Model 1; Altec 150W amp; SAE 130W amp; assorted mics; Beyer mic stands and booms; 100' mic snakes; Peavey stacks SP-1 & FH-1 and mini monitors: Ludwig 14 pc. drum set; Hammond B-3 organ; 825 Leslie; Fender Rhodes 88; Fender super 6 amp; Moog Opus 3 synthesizer. (605) 665-4169.

Wanted: Shure SR-107 10 Band eq in good to excellent shape. (313) 544-3046.

C-24, M49 and other old mics and used rec gear for sale (415) 441-8934 or (415) 527-6167.

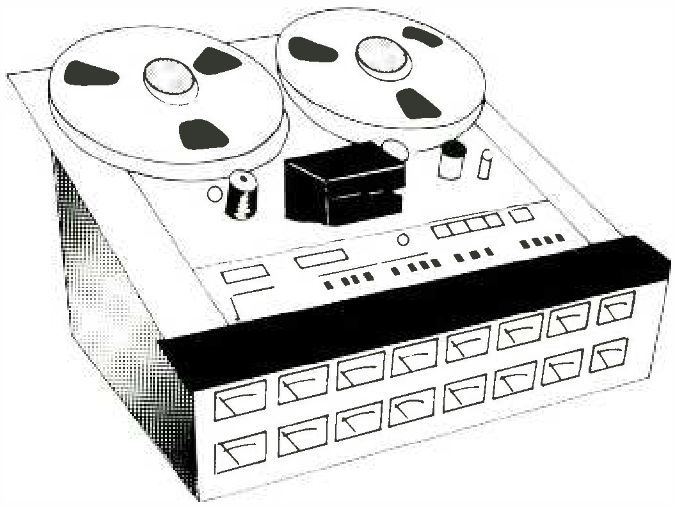
FREE CATALOG!! Complete Selection Of Consumer And Industrial Audio/Video Tape And Accessory Products. COMPETITIVE PRICES! PROMPT DELIVERY! Corporate/Bid Solicitations Welcome. Wyco Sales, P.O. Box 887, Cary, NC 27511. (919) 467-8113.

THE RECORDING BOOK THAT IS ROCKING THE MUSIC INDUSTRY

"Practical Techniques for the Recording Engineer" by Sherman Keene is endorsed by major Colleges, Universities, Recording Schools, Studios, Musicians and our Correspondence Students around the world. Recommended by reviewers of the MIX, Re/P, Guitar Player and other top publications.
THE BOOK: hard cover, 380 pages, 28 chapters (4 on computer assisted mixing), illustrations. $29.75 ($31.69 in Calif.) plus $2.50 for shipping.
THE CORRESPONDENCE COURSE: certificate course using two textbooks, corrected and graded homework, cassette dialog with the author via cassette, Basic, Intermediate and Advanced levels.
THE CURRICULUM: for schools only, Teacher's Manual (lesson plans for two textbooks, charts, suggested session content), Student's Workbook, Final Exams.
FOR INFORMATION OR TO ORDER contact:

S.K.P.
1626 N. Wilcox No. C-677
Hollywood, CA 90028

Order by phone using Visa or Mastercharge by calling (213) 708-2933.



SUBSCRIBER SERVICE

CHANGE OF ADDRESS

Planning to move? Please let us know six weeks in advance so you won't miss a single issue of MODERN RECORDING. Attach old label and print new address in space provided. Also include your mailing label whenever you write concerning your subscription to insure prompt service on your inquiry.

MODERN RECORDING
1120 Old Country Road
Plainview, NY 11803

Attach Label Here

New Address Here — *Please print*

Name ______

Address ______

City ______ State ______

Date ___/___/___ Zip ______

Introducing
Korg's KPR-77 Programmable Rhythmer

With these features, it really gives you rhythm, without the blues.

Until the KPR-77, if you wanted a full-featured rhythmer you had to spend over $1,000. So Korg developed a programmable at a price guaranteed to cheer you up. And the advanced micro-computer technology from their market-leading keyboards gives it all the features of the expensive units. Features like:

Programming Ease—Lets you program all instruments at the same time, in real or step time, simplifying the creation of more complex and realistic drum patterns. Advanced *LCD Display* gives you all the information you need, all the time.

Cassette Interface—For unlimited storage and fast retrieval of patterns, complete songs and full sets.

Greater Storage—Up to 48 two-measure patterns, and six 256-measure chains, which can be paired for 512-measure capability.

Edit Capability—Individual instrument editing in real or step time, and simplified chain edit functions.

8 Realistic Sounds—(including hand claps) Advanced circuitry creates incredibly realistic sounds comparable to far more expensive units. Includes bass drum, snare, open and closed high hat, cymbal, and high and low toms.

Plus the KPR-77 allows you to interface with synthesizers and other effects, offers programmable time signature, battery-protected memory, resolution up to 12 divisions per beat with a combined function for extended length patterns, selectable tom flam, mono and stereo outputs, programmable trigger outputs, sync in/out, seven channel mixer, accent and a whole lot more. It adds up to sound, features and price that really beat the blues.

For the name of your nearest Korg dealer, call (800) 645-3188 today. In N.Y., (516) 333-9100.

KORG ...we put it all together

Exclusively distributed by Unicord, Westbury, N.Y. 11590

For KPR-77 literature call or write: Unicord, 89 Frost St., Westbury, N.Y. 11590. For demo record and full Korg catalog write, enclosing $3.00.

CIRCLE 11 ON READER SERVICE CARD

INDUSTRY STANDARD
CHAPTER II

The New Otari 1/4" Two Channel 5050B-II

The best selling professional two track audio recorder has finally been improved. We've added the refinements you asked for: The inputs and outputs of the new B-II are transformerless, balanced. The elapsed time indicator is a real-time hours/minutes/seconds L.E.D. display–tape accurate at all speeds. The built-in oscillator provides both 1kHz and 10kHz calibration tones. And we added a low frequency adjustment to the reproduce equalizers.

Behind the clean, new look of the B-II are the same features, performance and reliability you expect from our famous 5050B. We didn't change the rugged quarter-inch thick deckplate or the cast aluminum frame. We kept the switch selectable NAB/IEC equalization, +4dBm/–10dBv output levels, half-track and quarter-track playback heads and three standard reference fluxivity levels. And, of course, the B-II still features three tape speeds, XL type connectors, front panel record equalization and bias adjustments, variable speed, "dump edit" function, and an integral splicing block.

The 5050B-II has been engineered like no other tape machine in the world. When you check out the specifications you'll know why we say it's the best $5,000.00 tape recorder available for under $2,500.00. When you work with it, you'll know that we've just raised the industry standard.

OTARI *Technology You Can Touch.*

Otari Corporation, 2 Davis Drive, Belmont, CA 94002
Tel: (415) 592-8311 Telex: 910-376-4890

CIRC 12 ON READER SERVICE CAR